

ESTIMATION OF REQUIRED BANDWIDTH FOR ORGANISATIONS

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Abstract: Over the years, organizations and individuals have suffered so much in the hands of the Internet Service Providers (ISPs) due to their inability to estimate the needed bandwidth. CiscoNet and Brad have proffered solution to adjust TCP window size and the round trip latency so that the desired bandwidth can be obtained. This paper examined the method of estimating bandwidth by combining the CiscoNet' and Brad's approaches.

Introduction

Bandwidth is a familiar term used in notable fields such as Computer Networks and Electronic Communication. In networks, Bandwidth is the amount of data that can be carried from one point to another in a given time period (usually a second). This kind of bandwidth is usually expressed in bits (of data) per second (bps). It is sometimes referred to as Throughput or data transfer rate - Occasionally, it is expressed as bytes per second (Bps). A modem that works at 57,600 bps has twice the bandwidth of a modem that works at 28,800 bps. In general, a link with a high bandwidth is one that may be able to carry enough information to sustain the succession of images in a video presentation.

In electronic communication, bandwidth is the width of the range (or band) of frequencies that an electronic signal uses on a given transmission medium. In this usage, bandwidth is expressed in terms of the difference between the highest-frequency signal component and the lowest-frequency signal component. Since the frequency of a signal is measured in hertz (the number of cycles of change per second), a given bandwidth is the difference in hertz between the highest frequency the signal uses and the lowest frequency it uses. A typical voice signal has a bandwidth of approximately three kilohertz (3 kHz); an analog television (TV) broadcast video signal has a bandwidth of six megahertz (6 MHz) -- some 2,000 times as wide as the voice signal.

In a packet network, the terms "bandwidth" or "throughput" often characterize the amount of data that the network can transfer per unit of time. Bandwidth estimation is of interest to users wishing to optimize end-to-end transport performance, overlay network routing,

and peer-to-peer file distribution. Techniques for accurate bandwidth estimation are also important for traffic engineering and capacity planning support. Existing bandwidth estimation tools measure one or more of three related metrics: capacity, available bandwidth, and bulk transfer capacity (BTC). Currently available bandwidth estimation tools employ a variety of strategies to measure these metrics.

It should be remembered that a real communications path usually consists of a succession of links, each with its own bandwidth. If one of these is much slower than the rest, it is said to be a bandwidth bottleneck.

In recent times, measurement of bandwidth required and usage has become a major problem. Thus, controlling them has been difficult. Most organizations pay much more than they require. Service Providers (ISPs) too, with the mind of maximizing their gains capitalizes on their clients' ignorance to cheat on them by making them pay more. What cannot be measured cannot also be controlled. Sometimes, even when an estimate of required bandwidth is gotten ISPs still sublet to other subscribers since it is always pretty difficult to fully utilize the required bandwidth at all times.

This paper looks into the problem of bandwidth measurement with a view to maximizing usage – having value for money in its true sense regarding networks. The major cause of network degradation was when the workload equates to or became greater than the Bottleneck capacity, leading to congestion across the WAN, which resulted to bottleneck. The overall effect was packet contention and high latency causing some packet to be loss or dropped for retransmission. All these accumulated to network degradation and poor performance in the quality of service delivered across the WAN. In 2009 alone, Nigerian banks achieved up to 30 percent cost savings, manufacturing Companies 25 percent and Schools about 20 percent cost saving on WAN recurrent costs through bandwidth optimization and management [1].

Definition of Terms

Bandwidth in computer networking refers to the data rate supported by a network connection or interface. One most commonly expresses bandwidth in terms of bits per second (bps). The term comes from the field of electrical engineering, where bandwidth represents the total distance or range between the highest and lowest signals on the communication channel (band).

Bandwidth represents the capacity of the connection. The greater the capacity, the more likely that greater performance will follow, though overall performance also depends on other factors, such as latency.

Latency is another element that contributes to network speed. The term *latency* refers to any of several kinds of delays typically incurred in processing of network data. A so-called *low latency* network connection is one that generally experiences small delay times, while a *high latency* connection generally suffers from long delays.

Latency vs. Bandwidth

Although the theoretical peak bandwidth of a network connection is fixed according to the technology used, the actual bandwidth you will obtain varies over time and is affected by high latencies. Excessive latency creates bottlenecks that prevent data from filling the network pipe, thus decreasing effective bandwidth. The impact of latency on network bandwidth can be temporary (lasting a few seconds) or persistent (constant) depending on the source of the delays (<http://compnetworking.about.com>)

Latency of Satellite Internet Service

Satellite Internet service illustrates the difference between latency and bandwidth on computer networks. Satellite Internet connections possess both high bandwidth and high latency. When loading a Web page, for example, most satellite users can observe a noticeable delay from the time they enter a Web address to the time the page begins loading. This high latency is due primarily to propagation delay as the request message travels at the speed of light to the distant satellite station and back to the home network. Once the messages arrive on Earth, however, the page loads quickly like on other high-bandwidth Internet connections (DSL or cable).

Besides propagation delays, latency also may also involve transmission delays (properties of the physical medium) and processing delays (such as passing through proxy servers or making network *hops* on the Internet).

Measuring Network Latency

Network tools like ping tests and traceroute measure latency by determining the time it takes a given network packet to travel from source to destination and back, the so-called *round-trip time*. Round-trip time is not the only way to specify latency, but it is the most common.

On DSL or cable Internet connections, latencies of less than 100 milliseconds (ms) are typical and less than 25 ms desired. Satellite Internet connections, on the other hand, average 500 ms or higher latency.

Related Works

Capacity bandwidth is a static baseline measure that applies over long time-scales (up to the time-scale at which network paths change), and is independent of the particular traffic dynamics at a time instant. Available bandwidth provides a dynamic measure of the load on a path, or more precisely, the residual capacity of a path. Additional application-specific information must then be applied before making meaningful use of either measure [7].

While measures of available bandwidth are certainly more useful for control or optimization of processes operating at short time scales, processes operating at longer time scales (e.g., server selection or admission control) will find estimates of both measures to be helpful. On the other hand, many network management applications (e.g., capacity provisioning) are concerned primarily with capacity bandwidth. Capacity bandwidth measure was used by them.

Dynamic changes in the routing or load can create errors in any measurement methodology. Unfortunately most currently available tools do not check for dynamic route or load changes during the measurement process [2].

Most large organizations obtain direct connections, which are leased lines that permit unlimited usage subject to the bandwidth of the line. Some customers purchase “dial-up” service which provides an intermittent connection, usually at much lower speeds. There are three major components: an annual access fee, an initial connection fee and in some cases a separate charge for the customer premises equipment (a router to serve as a gateway between the customer network and the Internet provider’s network). All of the providers use the same type of pricing: annual fee for unlimited access, based on the bandwidth of the connection [1]. A simulation of a sender and a receiver was used. A sender host transmits measurement packets to a receiver host, which immediately sends received packets back to the

sender host. The sender then estimates the available bandwidth of the path using the arrival intervals of the echoed packets. In every measurement, we use a *search range* to find the value of the available bandwidth [10]. Recent years have seen a surge in interest in available bandwidth estimation. A few tools have been proposed and evaluated in simulation and over a limited number of Internet paths, but there is still great uncertainty in the performance of these tools over the Internet at large [3]. Their research paper introduced Spruce, a simple, light-weight tool for measuring available bandwidth, and compares it with two existing tools, IGI and Pathload, over 400 different Internet paths. The comparison focuses on accuracy, failure patterns, probe overhead, and implementation issues. The paper verified the measured available bandwidth by comparing it to Multi-Router Traffic Grapher (MRTG) data and by measuring how each tool responds to induced changes in available bandwidth.

The measurements show that Spruce is more accurate than Pathload and IGI. Pathload tends to overestimate the available bandwidth whereas IGI becomes insensitive when the bottleneck utilization is large. The scale and complexity of the Internet makes the task of understanding and analyzing its properties extremely difficult. The diffuse style of administration and operation means that even such basic information as topology or link capacity is spread across multiple entities with little incentive to share it [6]. They described LinkWidth, a single-end controlled tool to measure the installed/bottleneck capacity and the available/un-utilized bandwidth. They give implementation details of how we extended the two existing techniques (Recursive Packet Train and Train of Packet Pair), to employ TCP RST packets sandwiched between TCP SYN packets. In addition, they show how to use a binary search approach to estimate installed/bottleneck and available/un-utilized capacity through a single tool.

For complex nature of corporate intranet networks and internet to comprehend the behaviour of these complex networks during data exchange [10] simulation method is useful. The developed algorithm for increasing the efficiency of data exchange in networks and the appropriate topology that suite the case showing increased efficiency without additional expenses.

TCP Throughput Calculation Formula

According to CiscoNet, Sometimes, we are feeling slow Internet connection, but don't know how we can measure the speed and what is right speed, download speed, TCP throughput, I can expect. Measuring and calculating TCP throughput is not that hard. See below famous TCP throughput formula [17]

RCV buffer size / RTT = Max TCP throughput = ? bps

**** Buffer size is normally 65Kbps**

For example,

$(64\text{Kbyte} \times 8\text{bit}) / 0.17 = 3011764 \text{ bps} = 3\text{Mbps}$,
(RTT=170ms)

RCV Buffer size / TCP receive window size

- RCV buffer size is denoted as TCP receive window size. Window systems have 64Kbyte of window size as default (Window NT and Millennium have 8Kbytes of window size). The TCP/IP standard allows for a receive window up to 65,535 bytes in size, which is the maximum value that can be specified in the 16-bit TCP window size field. Why 65Kbytes? Well, more accurate expression will be $65,535 = (2^{16}) - 1$. To improve TCP throughput, speed, performance whatever you called, in high speed connection or high delay network, you can increase TCP window size (reference RFC 1323). However, if transport link is not stable, it might give you worse performance. Packet loss or bottleneck in the network is the most likely factors that are leading to the TCP throughput reductions.

Optimal TCP window size

- Optimal RCV buffer size is considered to be $2 \times \text{BDP}$, where BDP is the Bandwidth*Delay Product

For example,

RTT is 20 ms, and connection speed is 10 Mbps.
 $2 \times (10\text{Mbps}/8 * .020\text{s}) = 50\text{Kbytes}$

Well, default window size 65kbytes is not adequate for today's network. In these days, most of network is 100Mbps or higher.

Round Trip Time(RTT)

- If you are not using TCP window scaling option (RFC 1323), TCP window size will be used as 64Kbytes. If then, Round Trip Time(RTT) is the main factor to decide TCP throughput between locations.

RTT 10 ms => TCP throughput = 52428000 bps
= 52Mbps
RTT 20 ms => TCP throughput = 26214000 bps
= 26Mbps

RTT 50 ms => TCP throughput = 10485600 bps
 = 10Mbps
 RTT 100 ms => TCP throughput = 5242800 bps
 = 5.2Mbps
 RTT 150 ms => TCP throughput = 3495200 bps
 = 4.3Mbps
 RTT 200 ms => TCP throughput = 2621400 bps
 = 2.5Mbps
 RTT 300 ms => TCP throughput = 1747600 bps
 = 1.7Mbps
 RTT 500 ms => TCP throughput = 1048560 bps
 = 1Mbps

** Used maximum TCP window size = 65Kbytes = 65535

This shows the deficiency with the greater RTT.

How to Calculate TCP throughput for long distance WAN links by BRAD HEDLUND

So you just lit up your new high-speed link between Data Centers but are unpleasantly surprised to see relatively slow file transfers across this high speed, long distance link — Bummer! Before you call Cisco TAC and start trouble shooting your network, do a quick calculation of what you should realistically expect in terms of TCP throughput from a one host to another over this long distance link.

When using TCP to transfer data the two most important factors are the TCP window size and the round trip latency. If you know the TCP window size and the round trip latency you can calculate the maximum possible throughput of a data transfer between two hosts, regardless of how much bandwidth you have.

Formula to Calculate TCP throughput

TCP-Window-Size-in-bits / Latency-in-seconds = Bits-per-second-throughput

So lets work through a simple example. I have a 1Gig Ethernet link from Chicago to New York with a round trip latency of 30 milliseconds. If I try to transfer a large file from a server in Chicago to a server in New York using FTP, what is the best throughput I can expect?

First let us convert the TCP window size from bytes to bits. In this case we are using the standard 64KB TCP window size of a Windows machine.

64KB = 65536 Bytes. $65536 * 8 = 524288$ bits

Next, lets take the TCP window in bits and divide it by the round trip latency of our link in seconds. So if our latency is 30 milliseconds we will use 0.030 in our calculation.

$524288 \text{ bits} / 0.030 \text{ seconds} = 17476266$ bits per second
 throughput = **17.4 Mbps maximum possible throughput**

So, although I may have a 1GE link between these Data Centers I should not expect any more than 17Mbps when transferring a file between two servers, given the TCP window size and latency.

What can you do to make it faster? Increase the TCP window size and/or reduce latency.

To increase the TCP window size you can make manual adjustments on each individual server to negotiate a larger window size. This leads to the obvious question: What size TCP window should you use? We can use the reverse of the calculation above to determine optimal TCP window size.

Formula to calculate the optimal TCP window size:

Bandwidth-in-bits-per-second * Round-trip-latency-in-seconds = TCP window size in bits / 8 = TCP window size in bytes

So in our example of a 1GE link between Chicago and New York with 30 milliseconds round trip latency we would work the numbers like this...

$1,000,000,000 \text{ bps} * 0.030 \text{ seconds} = 30,000,000 \text{ bits} / 8 = 3,750,000$ Bytes

Therefore if we configured our servers for a 3750KB TCP Window size our FTP connection would be able to fill the pipe and achieve 1Gbps throughput.

One downside to increasing the TCP window size on your servers is that it requires more memory for buffering on the server, because all outstanding unacknowledged data must be held in memory should it need to be retransmitted again. Another potential pitfall is performance (ironically) where there is packet loss, because any lost packets within a window requires that the entire window be retransmitted – unless your TCP/IP stack on the server employs a TCP enhancement called “selective acknowledgements”, which most do not.

Another option is to place a WAN accelerator at each end that uses a larger TCP window and other TCP optimizations such as TCP selective acknowledgements

just between the accelerators on each end of the link, and does not require any special tuning or extra memory on the servers. The accelerators may also be able to employ Layer 7 application specific optimizations to reduce round trips required by the application.

Reduce latency? How is that possible? Unless you can figure out how to overcome the speed of light there is nothing you can do to reduce the real latency between sites. One option is, again, placing a WAN accelerator at each end that locally acknowledges the TCP segments to the local server, thereby fooling the servers into seeing very low LAN like latency for the TCP data transfers. Because the local server is seeing very fast local acknowledgments, rather than waiting for the far end server to acknowledge, is the very reason why we do not need to adjust the TCP window size on the servers.

In this example the perfect WAN accelerator would be the [Cisco 7371 WAAS Appliance](#), as it is rated for 1GE of optimized throughput.

WAAS stands for: [Wide Area Application Services](#)

The two WAAS appliances on each end would use TCP optimizations over the link such as large TCP windows and selective acknowledgements. Additionally, the WAAS appliances would also remove redundant data from the TCP stream resulting in potentially very high levels of compression. Each appliance remembers previously seen data, and if that same chunk of data is seen again, that data will be removed and replaced with a tiny 2 Byte label. That tiny label is recognized by the remote WAAS appliance and it replaces the tiny label with the original data before sending the traffic to the local server.

The result of all this optimization would be higher LAN like throughput between the server in Chicago and New York without any special TCP tuning on the servers.

Formula to calculate Maximum Latency for a desired throughput

You might want to achieve 10 Gbps FTP throughput between two servers using standard 64KB TCP window sizes. What is the maximum latency you can have between these two servers to achieve 10 Gbps?

TCP-window-size-bits / Desired-throughput-in-bits-per-second = Maximum RTT Latency

524288 bits / 10,000,000,000 bits per second = **52.4 microseconds**

Calculating the Bandwidth of a file using Brad's Calculator

Brad's calculator readily estimates the bandwidth of a given file and the type of connection. For instance, if a file of 545kb is transmitted using a 14.4kbps it takes 5 minutes 10 seconds to be transmitted.

The Cisco's net aids in calculating throughput which is a basic element in determining bandwidth. Brad's calculator calculates bandwidth based on necessary input.

Combining these approaches gives an estimation of bandwidth required by an organisation.

Summary

Two key elements of network performance are bandwidth and latency. The average person is more familiar with the concept of bandwidth as that is the one advertised by manufacturers of network equipment. However, latency matters equally to the end user experience as the behavior of satellite Internet connections illustrates. Businesses use the term Quality of Service (QoS) to refer to measuring and maintaining consistent performance on a network by managing both bandwidth and latency in a coordinated fashion.

Conclusion

The combination of both the CiscoNet's and Brad's methods of calculating the Bandwidth and Throughput has given organizations a leeway to solve the bandwidth usage headache. Hence, requesting for increased bandwidth will no longer be necessary having known those other parameters like TCP window size and the round trip latency can be adjusted to obtain the desired bandwidth.

Future Research

As mentioned earlier, what cannot be estimated cannot also be controlled. Further researches can be conducted to determine appropriate controls necessary to ensure effective monitoring of the delivered bandwidth by the ISPs.

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