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An Empirical Observation Of 3G Networks Property:Bandwidth Analysis

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***Abstract*— The rapid and aggressive expansion in the field of telecommunication has been central to focus in ensuring the quality of service and provide data transfer services. Consistent internet connectivity was a key motivation to identify the parameters in defining the offered and available bandwidth in the specification of 3G network.We have done an extensive measurement study on multiple commercial 3G networks. We have investigated the performance of those 3G networks in terms of their data throughput, latency, packet loss rate, packet error rate and apply these parameters in delivering cloud computing infrastructure services primarily towards electronic governance programs of state and central government bodies. Their ability to provide service guarantees to different traffic classes under various loading conditions. . Our result will be useful to end users which empower people to decide which service provider is more reliable for them. It also shows the capacity varies widely not only across different operators but also across different measurement sites of the same operator**

Keywords: 3G, multi-tenancy, Throughput, Cloud, SaaS, TCP/IP measurement

I. INTRODUCTION

To know the 3G network characteristic which will provide the transparent technical report based on few parameters from the perspective of end users/customers.This study will be useful for telecommunication Industry. It tells them that yours network will need improvement or not with compare to other companies networks. We use network measurement to collect information for the following purposes:

- Evaluating performance of the network e.g. testing of new technologies and applications, throughput,,QoS performance metrics.
- Studying the properties of network - e.g. link characteristics, path, traffic.
- Generation of report like SLA (Service Level Agreement) validation
- Network operation assistance e.g. capacity planning, dimensioning, net- work monitoring and fault identification.
- Identify input for decision-making schemes e.g. routing, MPLS traffic engineering, CAC (Connection Admission Control).
- To compare the actual bandwidth and offered bandwidth offer by companies

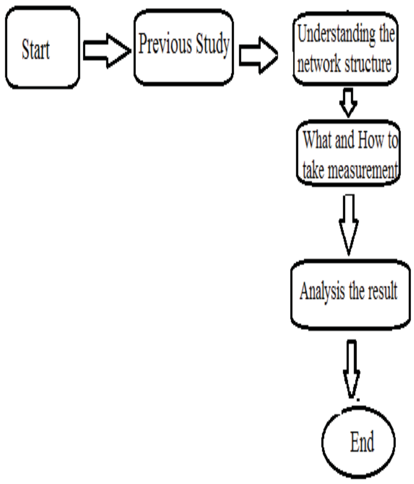


Figure 1

1.2-BASIC TOOL AND TERMINOLOGY

Their are some basic notions and terms related to computer network monitoring and auditing are listed. These terms are explained in the sense they are used in the network measurement scenario

1.2.1.Path:Links from source to destination in any network is called path. It is made of interconnected device on the way and their links.we can define path P of length m as the sequence of link (l1,l2,l3,. . . . .,lm) connecting node in between.

1.2.2Path Link Capacity:

Link capacity of a network vary with layers, each layer carry link capacity based on protocol .For example link capacity of transport layer is minimum of the link capacity of the of the path. So the capacity is the maximum amount of data that a link can support

1.2.3Latency:

Latency of network is defined by total delay for a bit of data to travel across the network between two end points. Different types of delay in the network are transmission delay, propagation delay, processing delay, queuing delay.

1.2.3.1Transmission delay: it include the time taken by the router to push the packet onto the network at the possible bit rate. Transmission delay is also known as serialization delay or store and forward delay.Transmission delay does not depend upon the path length or media ,it is the function of packet length and data rate of the link.

1.2.3.2Propagation delay: Propagation delay is the amount of time taken by signal to travel from sender to receiver or time taken by signal to travel by signal to travel from one transmission media to other. Propagation delay is the function of distance between transmitting and receiving media and speed of signal in the medium

1.2.3.3Processing delay: whenever packet arrives to the router, it checks its header to route on the outgoing link and also error checking takes place at each router so the processing delay is the sum of the delay caused by all the intermediate node processing the data . Processing delay is the key component in network delay. In the past, the processing delay has been ignored as insignificant compared to the other forms of network delay. However, in some systems, the processing delay can be quite large especially where routers are performing complex encryption algorithms and examining or modifying packet content. processing delay mainly occur on edge router of the network.

1.2.3.4Queuing delay: Queuing delay mainly depends upon the congestion of the network and buffer size of the router. Whenever a packet arrives to the router then it is processed and then transmitted , and router process one packet at a time so if incoming data rate is higher then transmitting rate then router puts the packet in the " router buffer" .so the queuing delay is directly proportional to the buffer size if buffer fills then router drops the packets. Queuing delay is the total time a packet spends inside routers on its way from the source node to the destination node.So total latency in the network is calculated by:

Ttotal = Ttrans +Tprop +Tprocessing + Tqueuing

Available Bandwidth:

Available bandwidth of a network is determined by the the total unused capacity of the link at a particular time.

Packet delay variation and inter-arrival time variation:Packet delay variation in the network is called as jitter. jitter is the variation in the time between packets arriving, caused by network congestion, timing drift, or route changes. A jitter buffer can be used to handle jitter. The delay is specified from the start of the packet being transmitted at the source to the end of the packet being received at the destination. Instantaneous packet delay variation is the difference in packet transfer delays for successive packets – this is what is usually called Jitter. Often Jitter is measured in terms of a time deviation from the nominal packet inter arrival times for successive packets.Voice packets in IP networks have highly variable packet-inter arrival inter- vals. Recommended practice is to count the number of packets that arrive late and create a ratio of these packets to the number of packets that are success- fully processed. You can then use this ratio to adjust the jitter buffer to target a predetermined, allowable late-packet ratio. This adaptation of jitter buffer sizing is effective in compensating for delays.Jitter and total delay in the network are totally different, having much of jitter in a network can increase the total delay in the network. This is because the more jitter you have, the larger your jitter buffer needs to be. Sometimes it is better to just drop packets or have fixed-length buffers instead of creating unwanted delays in the jitter buffers.Main reason of delay variation is the congestion in the network. Delay affects real time applications such as VoIP or video streaming where it causes breaks in audio and video.so to avoid this problem we use buffering, we first buffer the packet and then used after short delay,so that receiver get time to order the packet and streaming get continuous without any break.

II RELATED WORK:

Literature survey for the previous research work which have been carried out in analysis of 3G network on certain parameters is enumerated below.

2.1An Empirical Study on the Capacity and Performance of 3G Networks

The paper did an elaborated measurement study on multiple commercial third- generation(3G) networks. 3G network performance were investigated in terms of their data throughput, latency, video and voice call handling capacities, and their ability to provide service guarantees to different traffic classes under sat- urated and lightly loaded network conditions. Authors find out that the 3G network operators seem to have extensively customized their network configu- rations in a cell-by-cell manner according to the individual site's local demo- graphics, pro jected traffic demand, and the target coverage area of the cell. As such, the cell capacity varies widely not only across different operators but also across different measurement sites of the same operator. The results also show that it is practically impossible to predict the actual capacity of a cell based on known theoretical models and standard parameters, even with the support of key field measurements such as the received signal-to-noise ratio Ec =N0 .

2.2 3G/HSPA Performance in Live Networks from the End User Perspective

By introducing HSPA (High Speed Packet Access) to improve 3G networks, mobile broadband access is finally able to compete with fixed connections in performance regarding popular applications such as web browsing, VoIP and video. But it is not exactly clear how well the live networks behave. Authors filled this void by providing measurements in live 3G networks. They compared TCP and UDP goodput performance. Also one way delay and jitter measure- ment results are presented in stationary as well as in a mobile scenario. Authors have done the experiments under three measurements. The main measurement scenario consists of a stationary user in high signal strength conditions. The second scenario is also a stationary one, but in a considerably worse signal con- ditions. In addition, authors present a mobile user scenario, which is performed as a drive test in a freeway along a path with highly variable signal conditions. Before the actual tests, authors verified that there is full HSPA coverage in the used route.

3-Smartphone

Smartphone traffic represents an increasingly large share of Internet traffic but very little is known about its nature. Although studies have been done on smartphone traffic recently but they do now give the detailed comprehensive view of individual devices. Therefore the authors show their work in this field. They use packet sniffers on the smartphone devices for recording all sent and received packets. They use two data sets for the observation. First one consists of 10 users across windows mobile and android platform for which a logger is deployed which capture packet level traces. Second one consist of 33 Android users for which data regarding the bytes sent and received by each application in every 2 minutes are kept. The application is identified by the port num- ber. The data set has 1 to 5 months data of each user. It was found that the browsing contributes above 50% of the smartphone traffic while messaging, maps and media contributes 10%. The authors find that small transfers have many implications. The overhead of lower layer protocols was found to be high. In half of the transfers, header bytes constitute over 12% of the total bytes. In the presence of transport security, this overhead grows to 40% which includes SSL and all layers below it. This decreases the throughput of smartphone data transfers. Other factor affecting the throughput is packet losses. One of the reason for packet loss is that the downlink transfers are bottle- necked by the size of sender side transport buffer. This problem can be tackled by simply in- creasing the buffer sizes at servers that communicate with smartphone clients. It shows that current server-side transfer buffers are not well-tuned for smartphone workloads. The authors also study the interaction of smartphone traffic with the radio power management policy. They find that the current sleep timers, that is, the idle period after which the radio will go to sleep, are overly long. By reducing them based on current traffic patterns, radio power consumption can be reduced by at least 35% with minimal impact on performance. Therefore it again shows that current radio power management policies are not well-tuned for smartphone workloads.

III :AVAILABLE BANDWIDTH MEASURING TOOL

3.1Path load

It is active probing tool for estimating the available on the concept of "self-induced congestion". It estimate one-way delays of a periodic packet stream show increasing trend when the stream rate is larger than the avail- able bandwidth. The measurement algorithm is iterative and it requires the cooperation of both the sender and the receiver.

3.2Path Chirp: is a probing tool for estimating available bandwidth of network path. It uses an exponential flight pattern of probes, known as chirp . pathChirp is based on the concept of " self-induced congestion ,it rapidly increases the probing rate in each chirp and uses packet inter arrival times for estimation. We only use information of the relative delays between probe packets; so clock synchronization is not required between sender and receiver.

3.3WBest is a wireless bandwidth estimation tool designed for applications which are based on fast convergence time and low intrusiveness, such as multimedia streaming applications. WBest uses packet dispersion techniques to provide capacity and available bandwidth information for the underlying wireless net- works.

WBest is a two-stage algorithm:

1. A packet pair technique estimates the effective capacity over a flow path where the last hop is a wireless LAN (WLAN)
2. A packet train technique estimates achievable throughput to infer the available bandwidth.

The advantage of WBest is that it does not depend upon search algorithms to detect the available bandwidth but instead, statistically detects the available fraction of the effective capacity, mitigating estimation delay and the impact of random wireless channel errors.

Following assumptions are made in the best algorithm.

- Last hop of the wireless network is the bottleneck link on the whole net- work path. Here the bottleneck link means the last hop wireless network has both the smallest available bandwidth and the smallest capacity along the network path.
- There is no significant changes in network conditions between the two steps of the WBest algorithm.
- There is no overflow in the pat.

3.4Spruce:

(Spread Pair Unused Capacity Estimate) is a tool to measure available bandwidth. It analysis the arrival rate at the limiting hop(tight link) by sending pairs of packets spaced so that the second probe packet arrives at a bottleneck queue before the first packet departs the queue. Spruce then calculates the number of bytes that arrived at the queue between the two probes from the inter-probe spacing at the receiver. Spruce computes the available bandwidth as the difference between the path capacity and the arrival rate at the bottle- neck. Spruce treat capacity measurement from available bandwidth measurement differently. It assumes that capacity can be measured easily with one of the capacity measurement tools and that capacity stays stable when measuring available bandwidth.

Available Bandwidth Experiment and Methodology:

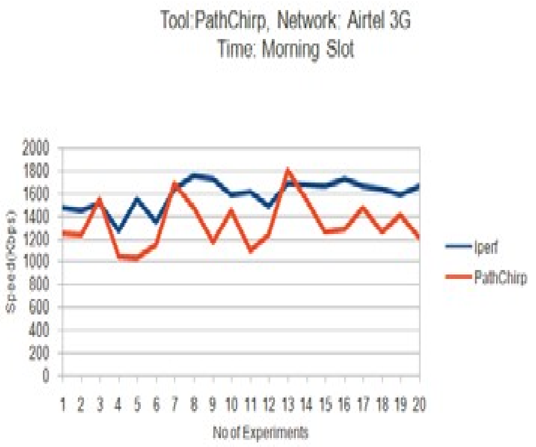
To determine the actual values of available bandwidth for comparison with those proposed by tools, one would need to setup for HSPA or EVDO networks. But then they would also suffer from the disadvantage of not being able to model real network well. We conducted our measurements on real networks. The celular were taken on HSPA(MTNL,Airtel,Idea) and MTS(EVDO). We used the following two setup :

1. we run parallel test for the Iperf and one of tool for 30 sec on two different machine.
2. we run Iperf and tool one after another for 30 sec.

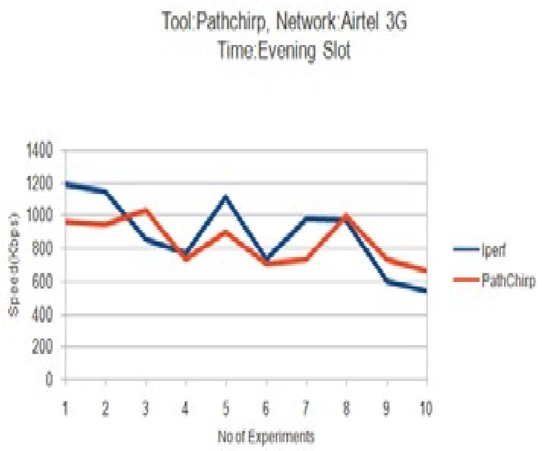
Both the measurements are taken without any gap which made the effect of diurnal patterns minimal. We estimate the range of values obtained from Iperf as ground value and compare it with those obtained from tool.

Available Bandwidth Analysis

We evaluate the performance of these tools on 3G network service providers like MTS, MTNL and Airtel. Additionally we use commercial networks as test bed and also propose a framework for measuring the maximum error possible in these tools on such environment. we choose two different time slot of 2 hours when we took all the measurement, one is morning time (3am - 5am) and evening slot time (7pm - 9pm).

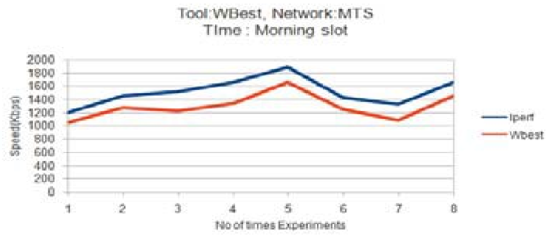


(a) Morning Time Analysis

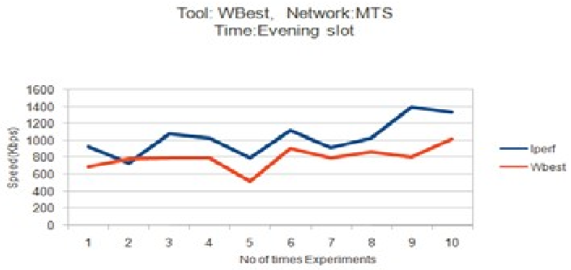


(b)Evening Time Analysis

Fig-2 Airtel 3G Available Bandwidth Analysis(a)(b)



(a)Morning Time Analysis



(b)Evening Time Analysis

Fig-3 MTS Available Bandwidth Analysis(a)(b)

- PathChirp and WBest give quite close results to Iperf estimates. We do not select Pathload as tool is known to be intrusive and hence not very suitable on cellular networks.We do not select Assolo, as it is an extension of PathChirp which employs a different kind of chirp profile.
- At morning 4 am, we can see that 90% of the predictions have less than10% of error. PathChirp when tested at this time gave an error of 17% if we considered the iperf estimates to be ground truth. Therefore the maximum possible error in the readings is 17+10 which is 27%.
- At the evening timing, we saw that 85% of the predictions have less than 19% of error. PathChirp when tested at this time gave an error of 21% if we considered the iperf estimates to be ground truth. Therefore the maximum possible error in the readings is 19+20 which is 39%.Therefore error in PathChirp estimates is bounded by 39% and 30% maximum error.
- The errors obtained above are just an upper bound and the actual error may be much smaller. But we are not able to estimate it correctly because the link is bursty and hence ground truth estimation becomes error prone leading to increase in total error.

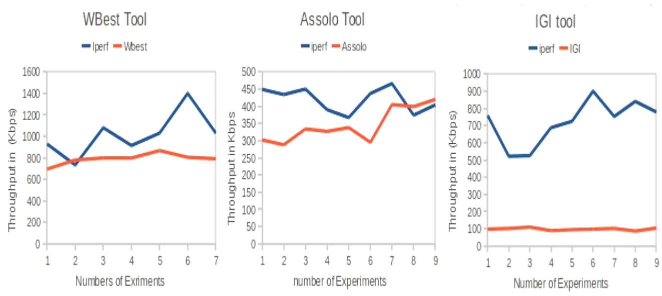
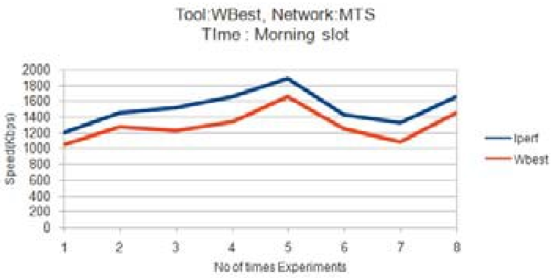


Figure -4Comparision Between Tool



CONCLUSION AND FUTURE WORK:

A more tough and thorough tests should be done in urban and rural field experiments. As we had measured the performance of five service provider, this experiment can be extended up to all the service provider present in the geo- graphic location where experiment is being performed. Our results are collected from few places only. So it can be extend into other cities where 3G network is available. We will also extend this work to other 3G devices like mobile. We can develop more robust measurement application suite which can be fit well with the internet network and particular to SaaS/ IaaS/ PaaS cloud computing resources audit with a facility of customized functions.

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