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Mobile Crowd-Sourcing Internet Measurement

Par

Pascal Ghobeira

Encadré par: Dr. Nicolas Rouhana

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Dr. Fadi Khoneisser
Dr. Nicolas Rouhana
Dr. Abed Ellatif Samhat
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ABSTRACT

The increase in Internet-enabled mobile devices provides an opportunity for a fine-grained observation of the performance of the Internet and a real time tracking of Internet quality of service. These mobile devices can operate as active monitors and probe the network to provide real time tracking of the Internet performance inside the network. The results produced by a large number of participants can be merged to obtain a detailed graph of the Internet.

With the deployment of high speed wireless networks in Lebanon, an increasingly number of users are taking advantage of the newly available services in cellular networks. However, mobile users still need a way to assess their internet performance based on their mobile operator to ensure that they are being provided a reasonable level of service quality.

In this thesis, we have researched the problems faced in IP networks in order to deduce the set of key performance indicators in mobile wireless networks. We propose a distributed collaborative platform to measure the key performance indicators across different Internet access points in order to continuously assess Internet connectivity and performance and have real time tracking of Internet Quality of Service using mobile devices and Location-Based services.

Alfa and Touch are two mobile operators in Lebanon currently deploying their Long-Term Evolution networks claiming that it could reach speeds of 100 or even 150 Mbps in some locations. Our project tackled different types of Internet measurements including FTP and HTTP tests using mobile terminals to obtain a map of coverage and Quality of service by mobile operator. The measurements obtained will be compared against the regulations issued by the Lebanese Telecommunications Regulatory Authority.

We have discovered that the LTE network of both operators was better compliant than their respective 3g networks with the regulations of the Lebanese Telecommunications Regulatory Authority regarding the values of the following metrics that were measured during our project from different locations and time of day: Round trip delay, Jitter, Ratio of packet loss.
1. **Introduction and Motivation**

With the deployment of high-speed wireless networks, mobile devices have seen a large growth in usage with an ever increasing number of applications being designed for these types of devices. Due to the mobility of users, the mobile devices operate in different environments with a frequent change in network performance.

Traditionally, network performance measurements have been done from within the cellular operator network by setting observation points on different network interfaces. Furthermore, operators rely on drive tests for service verification once the service has been launched, whereas monitoring user and service level performance regularly by drive tests alone is expensive and time consuming. As well, the performance perceived by the user can be affected by factors which are beyond the control and detection by the mobile operator observation points and drive tests.

The main goal of this thesis is to propose a distributed collaborative platform that collects measurements of several network characteristics related to performance for two mobile operators in Lebanon: Alfa and Touch. To simulate how the distributed platform will work, performance indicators including achievable TCP throughput, bandwidth, ratio of packet loss, jitter and, round trip delay were measured from different locations at different time of day. The Lebanese Telecommunications Regulatory Authority (TRA) has issued target levels for QoS and network performance for wireless broadband data services shown in table 1 below [10].

<table>
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<th>QoS/Network Performance Parameter</th>
<th>Target Level</th>
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<td>Service Availability</td>
<td>≥ 99.00%</td>
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<tr>
<td>Supply Time for Connection</td>
<td>95% completed on agreed day</td>
</tr>
<tr>
<td>Customer Fault Rate per leased line</td>
<td>1.25 failures per 100 customers per month</td>
</tr>
<tr>
<td>Fault Repair Time</td>
<td>95% within 24 hours</td>
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<td>Ratio of Packet Loss</td>
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<tr>
<td>Round trip Delay</td>
<td>≤ 95 milliseconds (ms) for national reference</td>
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<td>&lt; 250 milliseconds (ms) for International reference</td>
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Table 1 Target levels for QoS and network performance for wireless broadband data services
According to the regulations, the provider should have his wireless broadband data service available for 99% of the time. Also, 95% of new wireless broadband connection requests should be completed on the agreed day. For each operator, only 1.25 failures per 100 customers per month can be allowed and 95% of the faults should be repaired within the first 24 hours. Since we do not have access to internal data statistics of each operator, we cannot measure the aforementioned parameters. However, the table also includes target levels for three more parameters that we will measure: ratio of packet loss, round trip delay, and jitter.

Each mobile device taking part of the platform will collect a number of local measures that are then forwarded to a server, where they are assembled to generate a global map of the Internet for each operator. Mobility of nodes enables each single monitor to have different perspectives of the network, thus obtaining more detailed information. Moreover, location-based services available through mobile devices adds new dimensions to the analysis of the Internet where the measurement results can be mapped to a specific location at a specific time but for security reasons leaving the identity of the mobile device anonymous.

2. State Of The Art

2.1 LTE Network Overview

LTE or the E-UTRAN (Evolved Universal Terrestrial Access Network) is the access part of the Evolved Packet System (EPS). The main requirements for the new access network are high spectral efficiency, high peak data rates, short round trip time, and frequency flexibility.

![Figure 1 Network Solutions from GSM to LTE](image)

GSM was developed to carry real time services, in a circuit switched manner (in blue in fig.1), and with data services only possible over a circuit switched modem connection having very low data rates. The first step towards an IP based packet switched (in green in fig.1) solution was
made with the evolution of GSM to GPRS, using the same air interface and access method, TDMA (Time Division Multiple Access).

To reach higher data rates and data volume UMTS was developed with a new access network based on CDMA (Code Division Multiple Access). The access network in UMTS emulates a circuit switched connection for real time services and a packet switched connection for data services (in black in fig.1). In UMTS the IP address is allocated to the UE when a data service is established and released when the service is released. Incoming data services are therefore still relying upon the circuit switched core for paging.

The Evolved Packet System (EPS) is purely IP based. Both real time services and data services will be carried by the IP protocol. The IP address is allocated to the mobile when the mobile is switched on and the IP address will be released when the mobile switches off. The new access solution, LTE, is based on OFDMA (Orthogonal Frequency Division Multiple Access) to be able to reach even higher data rates and data volumes. High order modulation (up to 64QAM), large bandwidth (up to 20 MHz) and MIMO transmission in the downlink (up to 4x4) is also a part of the solution. The highest theoretical data rate is 170 Mbps in uplink and with MIMO the rate can be as high as 300 Mbps in the downlink.

The core network EPC is prepared to work with other access technologies not developed by 3GPP, like WiMAX and WiFi. Non 3GPP developed access solutions are divided in trusted and non-trusted. This division is not based on the technical solution but the business relation/agreement between the operators.

The LTE access network is simply a network of base stations, evolved NodeB (eNB), generating a flat architecture (figure 1). There is no centralized intelligent controller, and the eNBs are normally inter-connected by the X2-interface and towards the core network by the S1-interface (figure 2). The reason for distributing the intelligence amongst the base-stations in LTE is to speed up the connection set-up and reduce the time required for a handover. For an end-user the connection set-up time for a real time data session is in many cases crucial, especially in on-line gaming. The time for a handover is essential for real-time services where end-users tend to end calls if the handover takes too long.
Another advantage with the distributed solution is that the MAC protocol layer, which is responsible for scheduling, is represented only in the UE and in the base station leading to fast communication and decisions between the eNB and the UE. In UMTS, the MAC protocol and scheduling, is located in the controller and when HSDPA was introduced an additional MAC sub-layer, responsible for HSPA scheduling, was added in the NB.

The scheduler is a key component for the achievement of a fast adjusted and efficiently utilized radio resource. The Transmission Time Interval (TTI) is set to only 1 ms. During each TTI the eNB scheduler shall:

- **Consider** the physical radio environment per UE. The UEs report their perceived radio quality, as an input to the scheduler to decide which Modulation and Coding scheme to use. The solution relies on rapid adaptation to channel variations, employing HARQ (Hybrid Automatic Repeat Request) with soft-combining and rate adaptation.

- **Prioritize** the QoS service requirements amongst the UEs. LTE supports both delay sensitive real-time services as well as datacom services requiring high data peak rates. To schedule a low data rate, real-time service leads to a pleased customer but a low utilized radio spectrum.

- **Inform** the UEs of allocated radio resources. The eNB schedules the UEs both on the downlink and on the uplink. For each UE scheduled in a TTI there will be a Transport Block (TB) generated carrying user data. In DL there can be a maximum of two TBs generated per UE – if MIMO is used. The TB will be delivered on a transport channel. In LTE the number of channels is decreased compare to UMTS. For the user plane there is only one shared channel in each direction. The TB sent on the channel, can therefore
contain bits from a number of services, multiplexed together. In theory the highest number of users that can be scheduled during 1ms is 440, presuming 20 MHz band and 4x4 Multi User MIMO.

To achieve high radio spectral efficiency a multicarrier approach for multiple-access was chosen by 3GPP. For the downlink, OFDMA (Orthogonal Frequency Division Multiple Access) was selected and for the uplink SC-FDMA (Single Carrier - Frequency Division Multiple Access) also known as DFT (Discrete Fourier Transform) spread OFDMA (figure 3).

![Figure 3 OFDMA and SC-FDMA](image)

OFDM is a multicarrier technology subdividing the available bandwidth into a multitude of mutual orthogonal narrowband subcarriers. In OFDMA these subcarriers can be shared between multiple users. This solution is achieving very high spectral efficiency, but requires fast processors. It makes it possible to exploit variations in both frequency and time domains. The OFDMA solution leads to high peak-to-average power ratio requiring expensive power amplifiers with high requirements on linearity, increasing the battery consumption. This is no problem in the eNB, but would lead to very expensive handsets. Hence a different solution with lower requirement on the handset was selected for the UL.

To enable possible deployment around the world, supporting as many regulatory requirements as possible, LTE is developed for a number of frequency bands, ranging from 800 MHz up to 3.5 GHz. The available bandwidths are also flexible starting with 1.4 MHz up to 20 MHz. LTE is developed to support both the time division duplex technology (TDD) as well as frequency division duplex (FDD).

Since LTE provides high spectral efficiency, supports high data rates and implements flexible access architecture, it is proven to become a success amongst operators as well as customers. [1]
2.2 Internet Protocols

2.2.1 Transport Layer Protocols

**TCP**

The Transport Control Protocol is a connection oriented reliable transport-layer protocol, which means that two entities in a network first establish a connection between themselves, before transmitting the actual data. The main functionality is to deliver a fault-tolerant data transmission method over IP, by retransmitting data lost or corrupt on the way from sender to receiver. TCP has flow control, congestion control and guarantees reliable error free in-order delivery of data. Because of this feature, TCP does not work well with real time-applications such as streaming video, as it will require extra high amount of bandwidth required due to retransmission [2].

**UDP**

User Datagram Protocol is a simple connectionless best-effort transport-layer protocol. There are no mechanisms for error or flow control like in TCP, and therefore no guarantees that every packet is received, or that packets are received in the correct order. UDPs primary field of use is mainly real time applications such as multiplayer gaming, voice over IP and video streaming. Applications where low delays are highly appreciated, like DNS, are also utilizing UDP [3], [4].

2.2.2 Streaming Media

Streaming media is media (audio or video) that is being presented to the user while being delivered. This technology is widely used in the internet as a means for viewing movie clips or listening to an internet radio. Some TV-stations are also streaming some of their live-broadcasts out on the internet, and with the introduction of 3G, mobile phones can be used as a means of obtaining such a stream from the internet. Protocols used while streaming are RTSP and RTP over UDP [5].

2.2.3 Application Layer Protocols

**FTP**

File Transfer Protocol is a simple application-layer protocol designed for transferring files from a server to a client or in the opposite direction. FTP is run over TCP protocol and communication between the two data-exchanging parts is done by using one port for control and another for the data-transfer itself. [6]

By transferring a file of known size and recording the time it needed to be transferred, we can calculate the throughput using FTP by dividing the file size by the time needed.
HTTP

Hyper Text Transfer Protocol is a protocol developed for the World Wide Web. It is a stateless request/response protocol used primarily for web-browsing. It runs over TCP, and typically uses the common port 80 for request and data transfer. [7]

Although FTP is the standard for measuring user throughput, using HTTP to measure the throughput has its own advantages since the client does not need any special tools or applications to perform the measurements. Only a web browser which runs flash is needed. Below is a description how HTTP could be used to measure download:

1. Small binary files are downloaded from the web server to the client, the server then measures that download to estimate the connection speed.
2. Based on this result, the server chooses how much data to download for the real test. The goal is to pick the right amount of data that the client can download in 10 seconds, ensuring the server gets enough to be able to calculate an accurate result but not take too long.
3. The server prevents caches from throwing off results by appending random strings to each download.
4. Once the download has started, up to four HTTP threads are used to saturate your connection and get an accurate measurement.
5. Throughput samples are received at up to 30 times per second.
6. These samples are then aggregated into 20 slices (each being 5% of the samples).
7. The fastest 10% and slowest 30% of the slices are then discarded.
8. The remaining slices are averaged together to determine the final result.

Since the server is measuring data transported over HTTP (via Flash), there are the following factors that can affect speed: potential protocol overhead, buffering due to the many layers between the application and the raw data transfer, and throughput bursting due primarily to CPU usage. These factors lead the server to drop the top 10% and bottom 10% of the slices as outliers.

Additionally, the server should keep the default test length short for user experience. Because the test is shorter, the ramp-up period can take a significant part of the beginning of the test, leading the server to drop another 20% of the bottom result slices.
The Speedtest server can estimate the upload speed also as follows:

1. A small amount of random data is generated in the client and sent to the web server to estimate the connection speed.
2. Based on this result, appropriately sized chunk of randomly generated data is selected for upload.
3. The upload test is then performed in chunks of uniform size, pushed to the server-side script via POST.
4. Up to four HTTP threads can be used here as well to saturate the connection.
5. Chunks are sorted by speed, and the fastest half is averaged to eliminate anomalies and determine the result. [19]

2.3 Problems Faced in Packet-Switched Networks

In the telecom world, performance measurement is usually defined in terms of accessibility, retainability and quality. These terms work well for circuit-switched networks, where it is easy to identify network-measurable events and ultimately counters to each of these categories. Accessibility can be defined with the blocked calls, retainability with the dropped calls, and the quality with speech frame error rate. All these measured can be taken at the radio level (i.e. Base Station Subsystem, BSS) and can be easily translated into service quality.

However, for data services, the correlation between network metrics and user performance is not as direct as in the circuit-switched case. It is much more difficult to define counters in the BSS or other levels of the network that are able to capture the perception of service quality. There are two main reasons: first, data systems have many layers of protocols; i.e. a session may have accessibility problems, but at higher levels they may be seen just as increasing access delays. Secondly, radio data bearers are typically shared among different applications, thus mixing the information from very different sources; i.e. there exist services like Web browsing, which might be really degraded by delays, while some delays on background e-mail downloading would not be so much important. [11]

Therefore, it is important in wireless packet-switched networks that the key performance indicators measured provide a true indication about how each specific service is performing from the end-user point of view. In order to determine what key performance indicators are most significant to measure, we will look at different problems faced in packet-switched networks.

By inspecting the different problems faced in packet-switched networks: available bandwidth, end-to-end delay, jitter, and packet loss; we will deduce the metrics that should be measured by
the crowd-sourced framework of our project to be able to assess the Internet performance of the two mobile operators in Lebanon.

2.3.1 Available Bandwidth
The flow of packets is usually from source to destination through the best path. The maximum bandwidth of that path is equal to the bandwidth of the link with the smallest bandwidth. Lack of sufficient bandwidth will cause delay, packet loss and poor performance for the different applications that compete over the limited bandwidth of the links.

2.3.2 End-to-End Delay
The end-to-end delay is the sum of all the different delay types that affect the packets flowing from source to destination. Four of the important types of delay that make up end-to-end delay are as follows:

2.3.2.1 Processing Delay
Processing delay is the time it takes for a device to perform all the tasks needed to forward a packet from the input interface to the output interface.

2.3.2.2 Queuing Delay
Queuing delay is the amount of time that a packet spends in the output queue of an interface. Many factors related to the device characteristics affect the queuing delay.

2.3.2.3 Serialization Delay
Serialization delay is the time it takes to send all the bits of a frame to the physical medium for transmission across the physical layer.

2.3.2.4 Propagation Delay
Propagation delay is the time it takes for the bits of a frame to cross a physical link. This type of delay will be different across the different type of media used.

2.3.3 Jitter
The variation in delays experienced by the packets of the same flow is called jitter or delay variation. Packets of the same flow might not arrive at the destination at the same rate that they were released.
2.3.4 Packet Loss
Packet loss occurs when a network device has no buffer space on an interface to hold the new incoming packets and it ends up dropping them. In wireless links, packet loss can occur to other reasons as well, including radio links failures.

2.4 Network Performance Metrics
2.4.1 Achievable TCP Throughput
The achievable TCP Throughput is that amount of data per unit of time that TCP transports when in the TCP Equilibrium state.

TCP is connection oriented, and at the transmitting side, it uses a congestion window (TCP CWND). At the receiving end, TCP uses a receive window (TCP RWND) to inform the transmitting end on how many Bytes it is capable of accepting at a given time. Derived from Round-Trip Time (RTT) and network Bottleneck Bandwidth (BB), the Bandwidth-Delay Product (BDP) determines the Send and Received Socket buffer sizes required to achieve the maximum TCP Throughput. Then, with the help of slow start and congestion avoidance algorithms, a TCP CWND is calculated based on the IP network path loss rate. Finally, the minimum value between the calculated TCP CWND and the TCP RWND advertised by the opposite end will determine how many Bytes can actually be sent by the transmitting side at a given time.

Both TCP Window sizes (RWND and CWND) may vary during any given TCP session, although up to bandwidth limits, larger RWND and larger CWND will achieve higher throughputs by permitting more in-flight Bytes. At both ends of the TCP connection and for each socket, there are default buffer sizes. There are also kernel-enforced maximum buffer sizes. These buffer sizes can be adjusted at both ends (transmitting and receiving). Some TCP/IP Stack implementations use Receive Window Auto-Tuning, although, in order to obtain the maximum throughput, it is critical to use large enough TCP Send and Receive Socket Buffer sizes. In fact, they should be equal to or greater than Bandwidth Delay Product.
2.4.2 One-Way Delay (OWD)

One-way delay is the time it takes for a packet to reach its destination as shown in the figure 4 below.

![Figure 4 One-Way Delay](image)

When studying end-to-end performance, it is usually more interesting to look at the round-trip time and delay variation metrics which are derived from the one-way delay.

One-way delay between two nodes A and B can be measured by sending time stamped packets from A, and recording the reception times at B. The difficulty is that both nodes need clocks that are synchronized to each other. Since delay values will often be as low as the 100us to 10ms range, it will be important for node A and node B to synchronize very closely. GPS systems achieve synchronization to within microseconds. Ordinary application of NTP may allow synchronization to within several milliseconds, but this depends on the stability and symmetry of delay properties among those NTP agents used, and this delay is what we are trying to measure.
2.4.3 Round-Trip Delay

The round-trip time is the sum of the one-way delays from node A to node B and from node B to node A, and of the time it takes for node B to send the response.

Figure 5 Round-Trip Time

The weakness of measuring round trip delay instead of one-way delay can be summarized as follows:

- The Internet path from a source to a destination may differ from the path from the destination back to the source ("asymmetric paths"), such that different sequences of routers are used for the forward and reverse paths. Therefore round-trip measurements actually measure the performance of two distinct paths together.
- Even when the two paths are symmetric, they may have radically different performance characteristics due to asymmetric queuing.
- Performance of an application may depend mostly on the performance in one direction.
- In quality-of-service (QoS) enabled networks, provisioning in one direction may be radically different than provisioning in the reverse direction, and thus the QoS guarantees differ.

On the other hand, the measurement of round-trip delay has two specific advantages:

- Ease of deployment: unlike in one-way measurement, it is often possible to perform some form of round-trip delay measurement without installing measurement-specific software at the intended destination. A variety of approaches are well-known, including use of ICMP Echo or of TCP-based methodologies.
- Ease of interpretation: in some circumstances, the round-trip time is in fact the quantity of interest. Deducing the round-trip time from matching one-way measurements and an assumption of the destination processing time is less direct and potentially less accurate.
The round-trip time impacts the performance for window-based transport protocols such as TCP by influencing the achievable throughput at a given window size.

Furthermore, for interactive applications such as conversational audio/video, instrument control, or interactive games, the round-trip time represents a lower bound on response time, and thus impacts responsiveness directly.

The round-trip time is often measured with tools such as Ping which send ICMP Echo Requests to a destination, and measure the time until the corresponding ICMP Echo Response messages arrive.

While round-trip time reported by Ping is relatively precise measurement some network devices may prioritize ICMP handling routines, so that the measured values do not correspond to the real values.

2.4.4 Delay Variation (Jitter)

The Internet Engineering Task Force IP performance metrics working group (ippm) defines the delay variation in RFC3393 as a measure of transfer delay variation with respect to the previous packet. This particular metric only compares the delays experienced by packets of equal size based on the fact that naturally delay is dependent on packet size. Simply, the delay variation is the difference between the one-way-delay of the selected packets:

For Packet n,

\[ \text{IETF Delay variation (n)} = \text{OWD (n)} - \text{OWD (n-1)} \]

Example:

- Average transfer time: \( \tau \) = 10 ms
- Packet 2 is delayed in transit for an additional 5 ms
- Then two IPDV values will be affected.

\[ \text{Delay variation (2)} = \text{OWD (2)} - \text{OWD (1)} = 15 - 10 = 5 \text{ ms} \]
\[ \text{Delay variation (3)} = \text{OWD (3)} - \text{OWD (2)} = 10 - 15 = -5 \text{ ms} \]
\[ \text{Delay variation (4)} = \text{OWD (4)} - \text{OWD (3)} = 10 - 10 = 0 \text{ ms} \]

With a slight difference, the ITU-T study group 12 defines the delay variation as a delay with respect to a reference. The reference is usually the minimum delay.

For Packet n,

\[ \text{ITU-T Delay variation (n)} = \text{OWD (n)} - \text{Minimum [OWD (*)]} \]

In our example, and according to the ITU-T definition, we obtain:

\[ \text{Delay variation (1)} = \text{Delay variation (3)} = \text{Delay variation (4)} = 0 \]
\[ \text{Delay variation (2)} = 5 \]
Delay variation is an issue for real-time applications such as audio/video conferencing systems. A jitter buffer is usually deployed to eliminate the effects of delay variation. [17][18]

2.4.5 One-Way Packet Loss

Packet loss is the probability of a packet to be lost in transit from a source to a destination. The measurement of one-way loss instead of round-trip loss is motivated by the following factors:

- The path from source to destination may be different than the path from destination to source.
- Performance of an application may depend mostly on the performance in one direction. For example, a file transfer using TCP may depend more on performance in the direction that data flows, rather than the direction in which acknowledgments travel.
- In quality of service enabled networks, provisioning resources in one direction is different than provisioning in the reverse direction, and thus QoS guarantees differ. Measuring the paths independently allows the verification of both guarantees.

Packet losses have a big impact on end-to-end performance. Bulk data transfers usually require reliable transmission, so lost packets must be retransmitted. The TCP protocol assumes that losses are due to congestion and thus reduce their transmission rate. For real-time applications, it doesn’t make sense to retransmit lost packets. Audio/video codec provide a level of robustness to packet losses.

Packet loss can be actively measured by sending a set of packets from a source to a destination and comparing the number of received packets against the number of packets sent.

\[
\text{Packet Loss} = \frac{\text{sent ICMP requests} - \text{received ICMP requests}}{\text{sent ICMP requests of these same ICMP}} \times 100
\]

Network devices also contain counters for events such as checksum errors or queue drops, which can be retrieved through protocols such as SNMP. When this kind of access is available, this can point to the location and cause of packet losses.

2.5 Measurement Classification

Network measurement can be realized by using either an active tool or a passive tool.

2.5.1 Passive Measurements

Passive measurements do not inject traffic into the network but analyze the existing network traffic by collecting packets traversing a network link, or a switch/router device. The main advantage of passive measurements consists in not introducing any bias in the results of the measurements. The measurements can be used for showing network traffic in real-time or for analyzing the network traffic offline after saving the measured data. Intrusion detection and prevention systems are direct applications of passive measurements.[12]
Passive tools can be implemented in hardware or software to monitor interfaces or be used for statistics:

a) Hardware tools are plugged in derivation on the link also known as wiretapping to monitor the corresponding interface.

b) Software tools such as wireshark and tcpdump are used to monitor packets traversing network interfaces.

c) Software tools such as SNMP are used for providing traffic statistics of a device.

2.5.2 Active Measurements

Active measurement tools inject probing packets into the network and then measure the response. Getting close to true values estimations implies increasing the number of probes to accurately report path dynamics. However, large amounts of probing traffic can bias the results since it is overloading the network and changing its status. Being able to carry this type of measurements without relying on network support is a main advantage over passive techniques. Measurement methodologies vary from one active tool to another depending on which technique they use: sending ICMP echo, sending UDP packets with port unreachable, varying packets’ TTL and size, using path flooding and using TCP simulation.[12]

2.6 Measurement Applications

2.6.1 Iperf

Iperf is a tool for measuring TCP and UDP bandwidth performance, allowing the user to set various parameters that can be used for testing a network, or alternately for optimizing or tuning a network. Iperf reports bandwidth, delay jitter, datagram loss. Iperf has a client and server functionality, and can measure the throughput between the two ends, either in a unidirectional or bi-directional way. It is open source software and runs on various platforms including Linux, Unix and Windows.

- UDP: When used for testing UDP capacity, Iperf allows the user to specify the datagram size and provides results for the datagram throughput and the packet loss.
- TCP: When used for testing TCP capacity, Iperf measures the throughput of the payload. One thing to note is that Iperf uses 1024*1024 for megabytes and 1000*1000 for megabits.
Iperf is significant as it is a cross-platform tool that can be run over any network and output standardized performance measurements. Thus it can be used for comparison of both wired and wireless networking equipment and technologies. Since it is also open source, the measurement methodology can be scrutinized by the user as well. [9]

2.6.2 Windows Performance Monitor

Performance Monitor provides a visual display of built-in Windows performance counters, either in real time or as a way to review historical data. It features multiple graph views that enable you to visually review performance log data. Windows performance monitor can be used to measure all sorts of data traffic, like bytes received and sent per second at a specific interface. One can log the information to comma-delimited text files which then can be imported into Microsoft Excel for plotting.

3. Measurements and Results:

The goal of end-to-end testing is to perform tests that can indicate the performance as experienced from an end user. The tests will be done without any prior knowledge of the operators’ internal networks. The end-to-end measurements are performed using laptops connected to 3g and LTE of two cellular operators currently active in Lebanon: Alfa and Touch.

3.1 Overview

![Logical overview of the test architecture](image)

Figure 6 Logical overview of the test architecture
The clients are connected to the internet through 3g and LTE networks of the cellular operators currently active in Lebanon. A measurement server at Beirut IX is configured to run the following services:

a) Iperf Server  
b) FTP Server  
c) Speedtest Server (HTTP)

Beirut IX is used to exchange internet traffic between ISPs locally without the need to forward the traffic to the upstream international transit provider. According to the Beirut Internet Exchange website, major ISPs accept traffic from each other and agree to carry the other’s packets to their downstream destination point. The main advantages of Beirut IX are the decrease in international bandwidth cost for the ISPs and the improved quality of service for the traffic between the ISPs.

![Beirut IX Trace Route Example](image)

Figure 7: Beirut IX trace route example
Figure 7 shows the advantage of having a much lower latency when the traffic is passed through Beirut IX. This is easily proved by running a trace route twice, once via link 1 directly through Beirut IX and second via link 2 through the international upstream transit provider which had much higher latencies.

Another measurement server at Touch is configured only as an FTP server and used by Touch to perform FTP download and FTP upload operations to assess their network performance. A disadvantage of using the server located at Touch as a measurement point is that the server cannot be accessed when the client is connected through the Alfa network.

To measure the network performance of both cellular operators in similar test conditions, we focused more to do the measurements against the server present at Beirut IX as shown in table 2 below:

Table 2 Measurement tests

<table>
<thead>
<tr>
<th>Metric measured</th>
<th>Tool</th>
<th>Protocol</th>
</tr>
</thead>
<tbody>
<tr>
<td>Round trip delay</td>
<td>Ping</td>
<td>ICMP</td>
</tr>
<tr>
<td>TCP achievable throughput</td>
<td>Iperf</td>
<td>TCP</td>
</tr>
<tr>
<td>Available bandwidth</td>
<td>Iperf</td>
<td>UDP</td>
</tr>
<tr>
<td>Ratio of packet loss</td>
<td>Iperf</td>
<td>UDP</td>
</tr>
<tr>
<td>Jitter</td>
<td>Iperf</td>
<td>UDP</td>
</tr>
<tr>
<td>Reordering of packets</td>
<td>Iperf</td>
<td>UDP</td>
</tr>
<tr>
<td>FTP download</td>
<td>-</td>
<td>FTP/TCP</td>
</tr>
<tr>
<td>FTP upload</td>
<td>-</td>
<td>FTP/TCP</td>
</tr>
<tr>
<td>HTTP download</td>
<td>-</td>
<td>HTTP/TCP</td>
</tr>
<tr>
<td>HTTP upload</td>
<td>-</td>
<td>HTTP/TCP</td>
</tr>
</tbody>
</table>
3.2 Round Trip Delay

As stated before, the round trip delay is the time it takes for a packet to go from one host to another, and back again. It is usually measured with the Ping application which normally exists in most operating systems. Round trip delays between the client laptops and the server located at Beirut IX has been measured for the 3g and LTE networks for both operators, Alfa and Touch. The round trip delays have been measured numerously from different locations. For both operators, LTE networks produced better results than 3g networks. One possible reason for this is that the LTE networks are not yet widely used and overloaded as the 3g networks.

![Figure 8](image1.png) Round trip time in Touch 3g network in Hamra on June 27, 2013

![Figure 9](image2.png) Round trip time in Touch LTE network on June 27, 2013

Figure 8 and Figure 9 show two ping examples to find the round trip delay between our client and server. Although 3g and LTE networks are not symmetrical, uplink packets may travel different paths than downlink packets; however if needed, one way delays could be considered to be equal half the round trip delay.

According to the Lebanese Telecommunications Regulatory Authority, round trip delay should be smaller than 95ms for national reference and smaller than 250ms for international reference. We tested the round trip delay for national reference from different locations and operators’ networks. The Beirut IX server is considered as a local reference.

We averaged all the round trip delay measurements that have been done in each location and operator network over the course of this project. Table 3 below shows the result.
Table 3 Round trip delay between client and Beirut IX server

<table>
<thead>
<tr>
<th>Location</th>
<th>Alfa 3g</th>
<th>Touch 3g</th>
<th>Alfa LTE</th>
<th>Touch LTE</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ghazir</td>
<td>140 ms</td>
<td>133 ms</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>Ain el-Mreisseh</td>
<td>53 ms</td>
<td>160 ms</td>
<td>23 ms</td>
<td>16 ms</td>
</tr>
<tr>
<td>Hazmieh</td>
<td>174 ms</td>
<td>157 ms</td>
<td>27 ms</td>
<td>44 ms</td>
</tr>
<tr>
<td>Hamra</td>
<td>63 ms</td>
<td>121 ms</td>
<td>25 ms</td>
<td>18 ms</td>
</tr>
<tr>
<td><strong>Average Round Trip Delay</strong></td>
<td>107 ms</td>
<td>143 ms</td>
<td>25 ms</td>
<td>26 ms</td>
</tr>
</tbody>
</table>

Both average round trip delays in Alfa and Touch 3g networks did not comply with the TRA regulation of a maximum of 95 ms for national reference. However, their respective LTE networks performed better and complied with the regulation.

3.3 TCP Upload Throughput With Different TCP Window Sizes

This test was done using Iperf tool to experiment with different TCP window size to look at eventual differences in the achievable upload throughput. TCP window size is the definition of how much data is in transit in the network at any time. If the TCP window size is very small, the data sender will remain idle while the data is in transit, causing the overall performance to decrease. The correct TCP window size should be set according to the Bandwidth Delay Product (BDP).

In data communications, bandwidth-delay product refers to the product of a data link's capacity (in bits per second) and its end-to-end delay (in seconds). The result, an amount of data measured in bits (or bytes), is equivalent to the maximum amount of data on the network circuit at any given time, i.e. data that has been transmitted but not yet acknowledged. Sometimes it is calculated as the data link's capacity multiplied by its round trip time.

```
Ping 212.36.217.4 with 32 bytes of data:
Reply from 212.36.217.4: bytes=32 time=15ms TTL=57
Reply from 212.36.217.4: bytes=32 time=18ms TTL=57
Reply from 212.36.217.4: bytes=32 time=16ms TTL=57
Reply from 212.36.217.4: bytes=32 time=16ms TTL=57

Ping statistics for 212.36.217.4:
   Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
   Approximate round trip times in milli-seconds:
                         Minimum = 15ms, Maximum = 18ms, Average = 16ms
```

Figure 10 Round trip time (Touch network)
Round trip time from client to server is an average of 16ms (Figure 10). Theoretical bottleneck upload bandwidth in LTE 50Mbps depending on the frequency in the cell. From this we compute the bandwidth delay product:

\[
\text{Bandwidth Delay Product (BDP)} = 50 \times 1024 \times 1024 \text{ bit/s} \times 0.018 \text{ s} = 115.2 \text{ Kbyte}
\]

This gives us the theoretical TCP window size that should be set when connected through the Touch LTE network.

Figure 11 TCP throughput with various window sizes

Figure 11 is a graphical representation of the results. As the figure shows, increasing the window size to reach the theoretical value computed before had a dramatic increase on performance.

3.4 Topology Discovery

3.4.1 Touch network:

The attempt to discover the topology of Touch data network by using Traceroute only, proved to be unsuccessful. The use of Traceroute can be limited by ISPs which drop ICMP packets to avoid the discovery of the internal organization routing.
The second hop of the Traceroute result shown in Figure 12 above was the same for all the Traceroute measurements made when connected through both the 3g and LTE networks of Touch.

To overcome the issue of Traceroute ICMP being limited by the ISPs, a UDP version of Traceroute which sends packets to a high port number can be used. The destination receiving the packet with the highly number port will respond with an ICMP Port Unreachable.

3.4.2 Alfa network:

It became clear from the Traceroute measurement results that Alfa has multiple internet connections from different internet service providers (ISP) and is implementing some sort of load balancing between them. As well, inside the network of each ISP many load balancers could be present. The presence of load balancers could explains the out-of-order datagram results when doing the Iperf UDP tests.
In the Traceroute measurement results shown in the figures above, the first four hops addresses lie in the private address space revealing to us the internal hops of the Alfa network. A simple Whois search against the IP addresses of the fourth hop shown in figure 13, 14 and 15 prove that they belong to two different internet service providers in Lebanon. Therefore, it is clear that a load balancer is present inside Alfa’s network. A limitation of Traceroute may be caused by the load balancers inside the internet service providers networks since this tool is unable to detect whether the obtained IP addresses belong to this or that path. Paris Traceroute is a new version of Traceroute able to detect the presence of such load balancers by implementing heuristics to recover from their effects.

### 3.5 FTP Measurements

FTP is used to measure TCP user data throughput in uplink and downlink direction. To minimize the effects of TCP establishment and slow start, the size of the files used should be proportional to the type of network to be tested. According to a previous research, a file size value between 500 Kilobytes and 1 Megabytes was suggested for 2.5G systems.

In our tests, a file size of 5 Mbytes was used for 3G networks and a file size of 10 Mbytes for LTE networks. The file name was chosen to be different before each test to ensure that no
caching is affecting our results. Figure 16 shows a summary of all FTP measurements for both operators over the period of this project. FTP download and upload average throughput proved to be higher in both Alfa and Touch LTE networks than their 3g networks respectively. However, FTP upload throughput was greater than FTP download throughput in LTE networks.

![Average FTP Throughput in Mbps](image1)

Figure 16 Overall average FTP throughput in Alfa and Touch

Measurements done at Hazmieh location produced some unexpected results. In that particular location, FTP download throughput was faster in Touch 3g than in Touch LTE as shown in Figure 17 below. Further investigation revealed that the received signal from Touch LTE

![FTP Download Throughput in Kbps for Touch 3g and LTE](image2)

Figure 17 FTP download throughput in Hazmieh location on May 30, 2013 at 10:00pm
network was at its lowest at this location so naturally if FTP downloads in Touch 3g network yielded better results than in Touch LTE network at this location, the same should be said about FTP upload measurements. However, the FTP upload throughput measurements yielded the normally awaited results where Touch LTE network performed better than Touch 3g network.

Using Windows Performance Monitor application, we were able in Figure 18 to plot every second so we can see the actual throughput at each second and find the maximum peak values for FTP upload measurements performed at Hazmieh location for Alfa LTE network and Touch 3g and LTE networks. Two FTP upload measurements are made for each Touch 3g and LTE networks. Also, One FTP upload measurement is plotted for Alfa LTE. Comparing between Touch 3g and Touch LTE networks, the maximum peak values were obtained in the Touch LTE network as one would normally expect and even performed better than Alfa LTE. In these particular measurements, a larger file size for ftp upload measurement could be used if we were to calculate the upload throughput more precisely for LTE networks.

![FTP Upload throughput in Kbps](image)

**Figure 18 FTP upload throughput tests in Hazmieh on May 30, 2013 at 10:00pm**

For Touch LTE network, the maximum peak was approximately 12 Mbps whereas for Alfa LTE, it was 10 Mbps. For Touch 3g network, the ftp upload bit rate was similar to a constant bit rate if we neglect the slow start. The maximum peak achieved was 2.5 Mbps in this network.
3.6 Iperf UDP Upload Available Bandwidth Measurements

Iperf could be used in UDP mode for measuring the available bandwidth. Only short duration measurements in the range of 10 seconds should be done so as not to disturb other production flows. The goal of UDP measurements is to find the maximum UDP sending rate that results in almost no packet loss on the end-to-end path.

A practicable procedure to find the available bandwidth value is to start with UDP data transfers with a 10s duration and with intermediate result reports at one second intervals. If the measured packet loss values are below the threshold then a new measurement with slightly increased data rate could be started. This procedure of small UDP data transfers with increasing data rate should be repeated until the packet loss threshold is exceeded. The maximum data rate that caused packet losses below the threshold could be seen as a good measurement of the available bandwidth on the end to end path.

Iperf ran in server mode listening to port 5001 on the server located at Beirut IX. A laptop ran in client mode and generated at constant bit rate a UDP stream towards the server as upload data. The laptop connected Alfa’s and Touch’s 3g and LTE networks using the dongles provided to us. Iperf ran with the following parameters in server mode:

```
Iperf.exe -s -u -i 1
```

And in client mode:

```
Iperf.exe -c 212.36.217.4 -u -b<bandwidth>
```

In our tests, we were streaming UDP datagrams at different constant bit rates from the client (laptop) to the server in an upload manner. We repeated the tests for both types of networks, 3g and LTE.

```
C:\iperf>iperf -c 212.36.217.4 -u -b 20M

Client connecting to 212.36.217.4, UDP port 5001
Sending 1470 byte datagrams
UDP buffer size: 64.0 KByte (default)
[ 3] local 10.113.241.144 port 63348 connected with 212.36.217.4 port 5001
[ ID] Interval     Transfer      Bandwidth
[ 3] 0.0-10.0 sec  23.8 MBytes  20.0 Mbits/sec
[ 3] Sent 17007 datagrams
[ 3] Server Report:
[ 3] 0.0-10.0 sec  23.8 MBytes  20.0 Mbits/sec  1.597 ms  0/17006 (0%)
```

Figure 19 Iperf UDP client
Figure 19 shows an example of the Iperf client configured to generate UDP traffic at a rate of 20 Mbps for 10 seconds. The server detects UDP datagram loss by ID numbers in the datagrams. Usually a UDP datagram becomes several IP packets. Losing a single IP packet will lose the entire datagram. To measure packet loss instead of datagram, the datagrams should be small enough to fit into one single packet. In our tests, Datagrams of 1470 bytes were used, this makes one datagram fit into one IP-packet, and a datagram loss is the same as packet loss. A total amount of 23.8 Mbytes were generated in the network. Figure 20 shows the results collected at the server located at Beirut IX at a 1 second interval indicated by the –i 1 parameter. We notice that the server received all the packets without any loss and was receiving at the same rate of the sender. The jitter values were also very acceptable for any type of VOIP and voice/video streaming applications. The total size of the data that was generated as uplink traffic during this test was 23.8 Mbytes. This particular measurement test took place at Ain el-Mreisheh location on June 24, 2013 at 5:30pm.

<table>
<thead>
<tr>
<th>Server listening on UDP port 5001</th>
</tr>
</thead>
<tbody>
<tr>
<td>Receiving 1470 byte datagrams</td>
</tr>
<tr>
<td>UDP buffer size: 108 KByte (default)</td>
</tr>
<tr>
<td>-----------------------------------</td>
</tr>
<tr>
<td>[ 3 ] local 212.36.217.4 port 5001 connected with 146.185.34.225 port 63348</td>
</tr>
<tr>
<td>[ ID ] Interval</td>
</tr>
<tr>
<td>[ 3 ] 0.0- 1.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 1.0- 2.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 2.0- 3.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 3.0- 4.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 4.0- 5.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 5.0- 6.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 6.0- 7.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 7.0- 8.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 8.0- 9.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 9.0-10.0 sec</td>
</tr>
<tr>
<td>[ 3 ] 0.0-10.0 sec</td>
</tr>
</tbody>
</table>

Figure 20 Iperf UDP server result

The Iperf client was configured to generate traffic at different bit rates but we made sure that the loss ratio does not exceed 10 or 15 percent. According to the Lebanese Telecommunications Regulations Authority, TRA, the ratio of packet loss inside wireless broadband data services should not exceed the 5% limit or otherwise the performance of different applications especially VOIP will degrade to an unacceptable level. Jitter values were acceptable and smaller than the TRA regulation. According to the UDP protocol, if two messages are sent to the same recipient, the order in which they arrive cannot be predicted. Therefore, out of order datagrams is a major issue in applications which use the UDP protocol to send data. In Touch 3g network, we had a large number of out of order datagrams which will be interpreted as lost packets by higher layers.
Figure 21 below shows that the average upload throughput at all locations measured using Iperf UDP test was higher for Touch 3g network.

![Average Iperf UDP Upload Throughput](chart1)

**Figure 21** Average Iperf UDP upload bandwidth in Mbps

Figure 22 below shows that the average jitter values in Touch’s 3g networks were much lower than Alfa’s 3g network.

![Average Jitter values](chart2)

**Figure 22** Average Iperf UDP jitter values in milliseconds
In Figure 23 above, we plotted the average packet loss ratio in both Touch 3g and Alfa 3g networks. The measurement packets traveled from Iperf client to Iperf server through multiple paths causing many out of order datagrams in Touch’s 3g network which may be interpreted as loss because UDP packets are not reassembled.

We also performed the Iperf UDP upload measurements to assess as well the performance of LTE networks of Alfa and Touch as well.

To avoid any misconception, we used to the same parameters in Iperf to test the two operators LTE networks at Hamra location. We generated 10, 15, 20 and 30 Mbps from the client and reported the results at the server. Figure 24 below shows the reported UDP upload throughput at the server.
Except when the client was set up to generate traffic at a bit rate of 20 Mbps, results proved that Alfa performed better than Touch in terms of upload bandwidth as shown in figure 24. However, jitter values, shown in Figure 25, were also higher for Alfa’s LTE network. No out of order datagrams, shown in Figure 27, were reported in Touch’s LTE network whereas many out of order datagrams were reported in Alfa’s LTE network which could be due to multiple paths.

![Jitter Values](image1)

Figure 25 Comparison between Alfa and Touch LTE UDP upload jitter values in Hamra location

![Packet Loss ratio %](image2)

Figure 26 Comparison between Alfa and Touch LTE UDP upload packet loss ratio in Hamra location
3.7 Iperf TCP Throughput Upload Measurements

End-to-end TCP measurements using Iperf can be used by users to detect throughput problems like unexpected slow file transfer time. However, an issue between the end system’s TCP stack could also be experienced by the end user as a slow file transfer time. The knowledge of this network metric is therefore important for distinguishing between issues with the end system’s TCP stack, or network related problems.
Figure 28 shows an Iperf TCP client with TCP window size of 128 Kbyte connecting to the server at Beirut IX and being able to transfer 26.8 Mbytes of data during this TCP session having the default duration of 10 seconds. The server shown in Figure 29 reported at intervals of 1 second specified by the –i parameter. The bandwidth reported for this TCP session is 22.1 Mbps and 26.8 Mbytes were transfer as an upload to the server. This particular measurement was for Touch LTE network at Ain el-Mreisseh location on June 24, 2013.

3.8 Iperf TCP test, Speedtest and FTP measurement results

According to the European Telecommunications Standards Institute, ETSI, the specific protocol used to perform a measurement may have an impact on the measurement results. For example, performing application layer measurements in two different mobile networks using both FTP and HTTP can result in a situation where one network produces better results for FTP measurements while the other one produces better HTTP results [14]. Figures 30 and 31 plot the Speedtest and FTP download measurement results in Alfa’s LTE and Touch’s LTE networks.
In Touch’s LTE network (Figure 31) the measurements showed that the Speedtest download throughput was higher than the FTP download throughput for all locations. In the upload tests, we also used Iperf tool as mentioned before to do the measurements. Figure 32 and 33 plot the Speedtest, FTP and Iperf TCP Upload measurement results in Touch’s and Alfa’s LTE network respectively.

Figure 31 Speedtest and FTP download measurement results in Touch LTE network

Figure 32 Speedtest, FTP and Iperf TCP Upload measurement results in Touch LTE network
As shown in Figure 32 and 33, Iperf reported higher upload TCP throughput. Also, the Iperf tool injected high amount of traffic into the network which could be the reason it had the best results since the effects of TCP establishment and slow start were minimized.

Figures 34 and Figure 35 shows the comparison between the three types of measurement tests in the 3g networks of the operators. HTTP performance proved to be better at low throughputs because the file transmitted between client and server were of small size whereas in the LTE networks a smaller file size could not have a much significant measurement result because the effects of TCP establishment and slow start are not minimized.
Figure 35 Speedtest, FTP and Iperf TCP upload measurement results in Touch 3g network

Similar to the upload tests in the 3g network of both operators, HTTP performance was better than FTP performance in the downlink as well as shown in Figure 36 and Figure 37

Figure 36 Speedtest and FTP download measurement results in Alfa3g network

Figure 37 Speedtest and FTP download measurement results in Touch3g network
4. **Conclusion and Further Work**

Using a mobile crowd-source approach, we measured selected performance indicators in the packet switched networks of Alfa and Touch, the two mobile operators in Lebanon. HTTP and FTP as well as the Iperf tool were used to measure the TCP throughput between the server located at Beirut IX and the mobile clients. Jitter, ratio of packet loss and round trip delay were also measured and compared to the regulations of Telecommunications Regulatory Authority (TRA).

<table>
<thead>
<tr>
<th></th>
<th>Alfa 3g</th>
<th>Touch 3g</th>
<th>Alfa LTE</th>
<th>Touch LTE</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ratio of packet loss</td>
<td>95%</td>
<td>30%</td>
<td>35%</td>
<td>40%</td>
</tr>
<tr>
<td>Jitter</td>
<td>30%</td>
<td>100%</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>Roundtrip Delay National Reference</td>
<td>15%</td>
<td>10%</td>
<td>100%</td>
<td>100%</td>
</tr>
<tr>
<td>Roundtrip Delay International Reference</td>
<td>90%</td>
<td>90%</td>
<td>100%</td>
<td>100%</td>
</tr>
</tbody>
</table>

Table 4 shows how many measurement results were compliant to the regulations of the TRA. While the LTE networks of both operators was compliant to these regulations, the 3g networks failed at many locations to operate within these boundaries.

Having proved before different types of measurement tests produced different results based on the test location and time of day, it is then a necessity to create a large scale crowd-sourced collaborative approach to identify the quality of service as experienced by mobile users. Using the results obtained, a map will be plotted containing coverage areas and service performance points for different operators.
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