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Measurement of GPRS Performance over Libyan GSM Networks: Experimental Results

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ABSTRACT

The General Packet Radio Service (GPRS) is a new bearer service for GSM that greatly simplifies wireless access to packet data networks, such as the Internet, corporate LANs or to mobile portals. The aim of this work is the measurement of the quality of service (QoS) parameters of GPRS over the Libyan GSM networks, Libyana mobile phone and Al-Madar Al-Jadeed Company. The measured parameters of GPRS are the throughput, round trip-time, delay time, packet loss, packet duplicate, upload speed, and download speed. To evaluate these Parameters, End-to-End measurements are used. At one end is the client (mobile) and at the other end is the measurement server. This server is located with interne address. A special analysis algorithm was implemented.net and used for analysis the measured data. Finally, the measured values of quality of surface parameters of GPRS over the two Libyan mobile operators are illustrated and compared with the theoretical values that could be calculated beforehand.

Keywords: General Packet Radio Service, Quality of Service, Data Performance, Signal Quality, Radio Frequency Performance.

I. INTRODUCTION

The General Packet Radio Service (GPRS), developed by the European Telecommunications Standards Institute (ETSI) [1]. It applies a packet radio standard to transfer user data packets in well-organized way between Mobile Stations (MS) and external packet data networks[1].The nature of wireless links is quite different compared to wire line networks; their latency and error prone characteristics make it a challenging environment for providing efficient transport. The GSM system can only support data services up to 9.6kbit/s circuit switched. The GPRS can improve this bit rates. However, commercial GPRS systems will be able to support rates up to 115kbit/s [2].

These provided GPRS services are affected with loss of packets during transmission over wireless network. The Packet losses may occur in the wireless environment more often than in wire line networks because of multiple reasons. The congestion traffic and surrounding buildings may cause interference resulting in packet losses as well as the hand offs in cellular wireless networks. Such conditions can also cause excess delays as the radio link layer may locally retransmit the corrupted segments [3,4].

This paper presents a system that measures the parameters of Quality of Service (QoS) over the two Libyan GSM networks (libyanna and AL-Madar Al-Jadeed), which have effect on the data transmission over these networks. The investigated Parameters are throughput, round-trip time, download and upload. The QoS can be classified as subjective and objective QoS.

II. GPRS MEASUREMENT PRINCIPLE

GPRS measurements are divided into three categories: data performance, signal quality, and Radio Frequency RF performance as shown in Fig.1.

- The Data Performance category emphasizes data-transfer-quality measurements (as perceived by customers) and GPRS layer-specific measurements.

- The Application layer measurements are used to evaluate the parameters directly perceived by the user (such as throughput and delay).

- The GPRS layer measurements offer insight about events on the GPRS layers that can impact the application layer performance. The measurements are made using a test mobile connected to a computer to trace data packets.

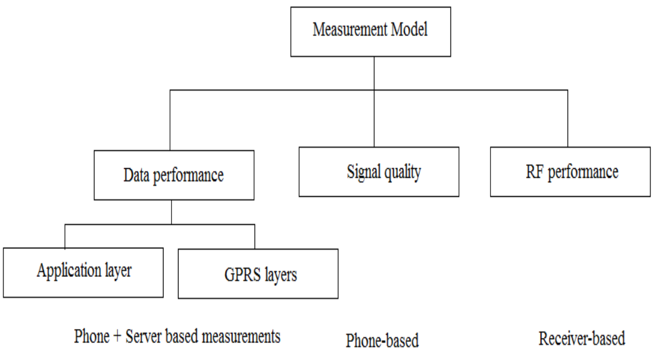

Figure.1: GPRS measurements model

Data performance at the application layer is measured end-to-end, which can be described as follows: One node (mobile) transmits data and another node (server) receives the data and measures its performance. The server can be a located at the Internet world. Since the measurements are made end-to-end, in the uplink the server measures the data received from the mobile and sends back the results. The ultimate objective at the application layer is to get the user perspective.

A. The Principles of QoS Measurement

Measuring network quality of service (QoS) is basically very close to network traffic measurements. In network traffic studies, the main point is the effects of the traffic on the network: network load, queuing performance, source traffic processes, large scale traffic flow models. Especially the analysis of traffic processes and models requires accurate information of collected traffic [6]. In QoS analysis, the network traffic itself is not the interesting thing, but it is rather just used as a tool to reveal the performance characteristics (delay, throughput, etc.) of networks. The measurement of QoS is divided in to three separated functional entities, measurement point, traffic measurement tool, and QoS analysis tool, as shown is Fig 2.

The actual measurements are done at measurement points that are, in practice, some network nodes under interest (e.g., routers, terminals…). General rule is that the more measurement points, the more accurately the network behavior can be determined, while the analysis also gets more complicated [7].

The QoS analysis tool is used to analyses the data provided by traffic measurement tools and calculates the actual QoS metrics. In practice, this means, for example, delay calculation of single packets traveling through measurement points. There usually exists a single QoS analysis entity in the measurement system, but naturally the actual calculation process can be distributed in nature.

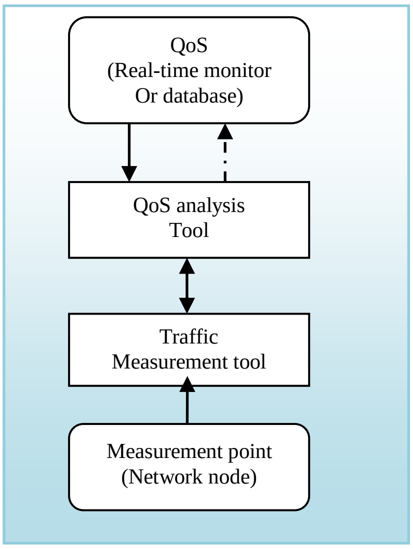

Figure 2: The basic principle of QoS measurements

B. End-to-End performance measurements

The end-to-end performance measurements can be executed as single-point measurements directly from the terminal, which uses some service as shown in Fig.3. The measurement software can be an application on top of the protocol hierarchy, in which case the measurement gives directly an idea about the application layer performance that is usually the desired case in QoS.

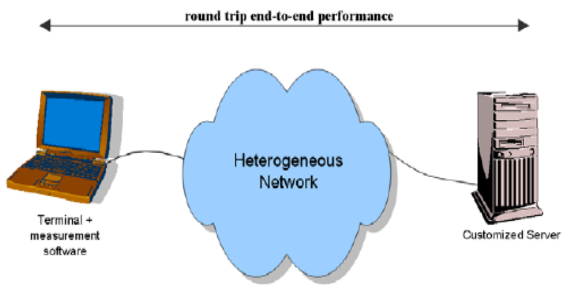

Figure 3: The single-point end-to-end performance measurement

As it can be assumed, with this kind of measurement setup, it is only possible to obtain information about the total round trip performance of the system. One way to enhancement the end-to-end performance measurement is to attach measurement software to both ends of the link. Then the performance of different directions (from terminal to server and from server to terminal) can be analyzed separately in addition to the total round trip performance.

III. SYSTEM ARCHITECTURE

The system is developed using client server architecture (end-to-end measurement). One end node is the client and the other is a measurement server as shown in Fig. 4. This server can be a located at the Internet world. The end-to-end measurements can be described as follows: One node transmits data and another node receives the data and measures its performance.

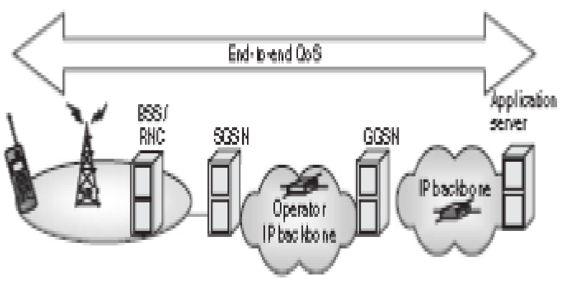

Figure 4: Block diagram of developed system

Two different computers are used, one was used as end hosts and the other was running as a client on the GSM network. Time of the client computer is equipped with a 2.8 GHz Intel (Due Core Centreno processor) and a 1GB RAM. The end host is equipped with 2.0 GHz Intel Pentium running Windows XP service Pack2 as its operating system and a 1GB RAM. As the measurements are done in real- time, it is important that the whole measuring environment does not produce wrong results. Therefore, these computers dedicated for measurements purpose only and there were no CPU consuming applications running, on them during the measurements.

Since the measurements are made end-to-end, in the uplink the server measures the data received from the mobile and sends back the results. In the downlink the measurements are made by the same software that generated the uplink data. For end-to-end throughput measurements, special software is used to generating bulk data transfers over TCP. For measuring latency a standard ping program can be used [5].

IV. MEASUREMENTS AND ANALYSIS

A. Packet Drop Test

In this test, the packet drops during transmission over wireless network is measured. First, the UDP protocol at transmitter is used to send many packets over GPRS network to server. Every packet represents identified record from database that is prepared for this test. At the receiver side we checked every delivered record. If it is not received at certain time we mark it as lost. If the packet correctly received at the receiver we acknowledge the transmitter. If it is received many times, we mark it as a duplicate in the table. The algorithms that implemented to perform these tasks are shown in flowcharts of fig. 5 and fig. 6 respectively.

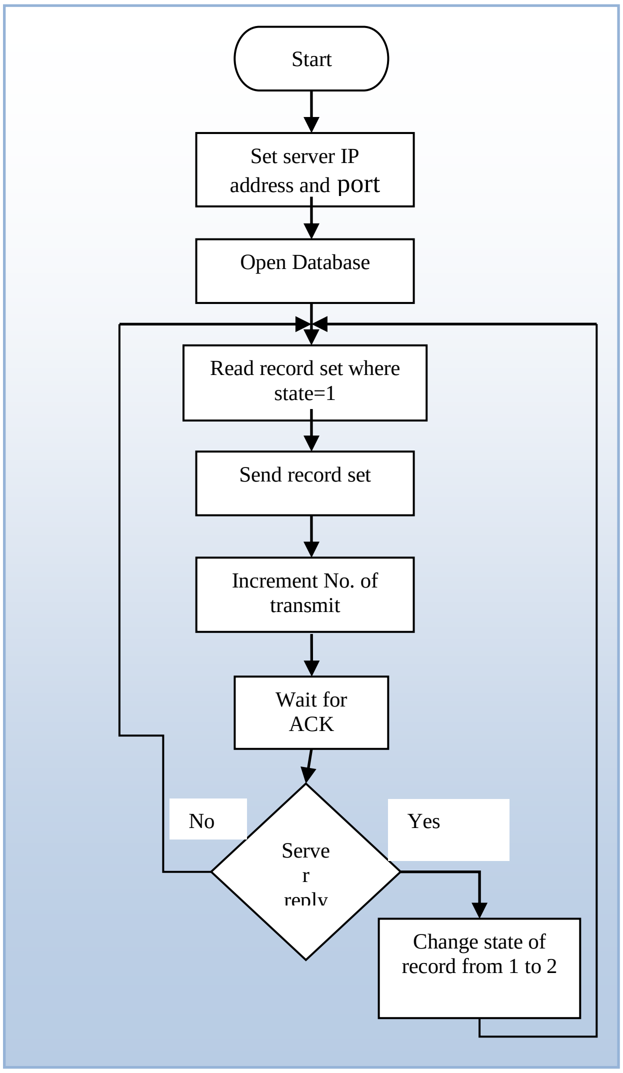

Figure 5: Flow chart of Transmitting Program (client)

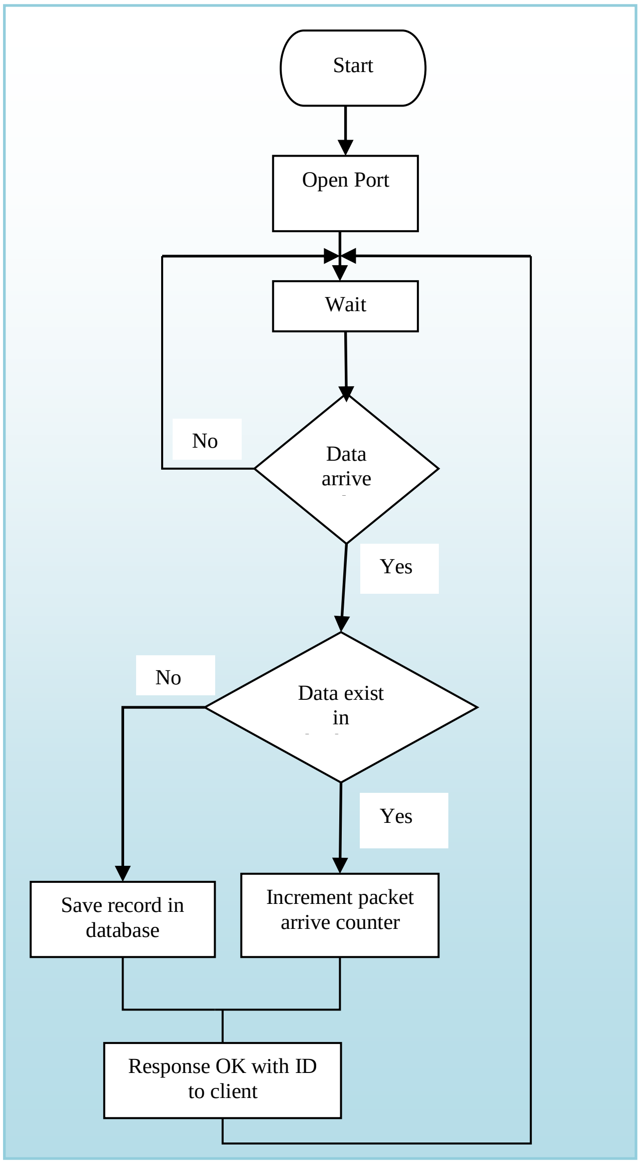

Figure 6: Flow chart of Receiving Program (Server)

B. Reliability Test

In this test the number of times disconnection that may occurs in GPRS is measured. To accomplish this we wrote two programs, one for client and other for server. We used WinSocket tool to make connection between them. After we read the state of connection to check if it is still connected, we record the time and state of the connection in database to know when the disconnection occurs. Fig. 7 shows the flowchart of this program.

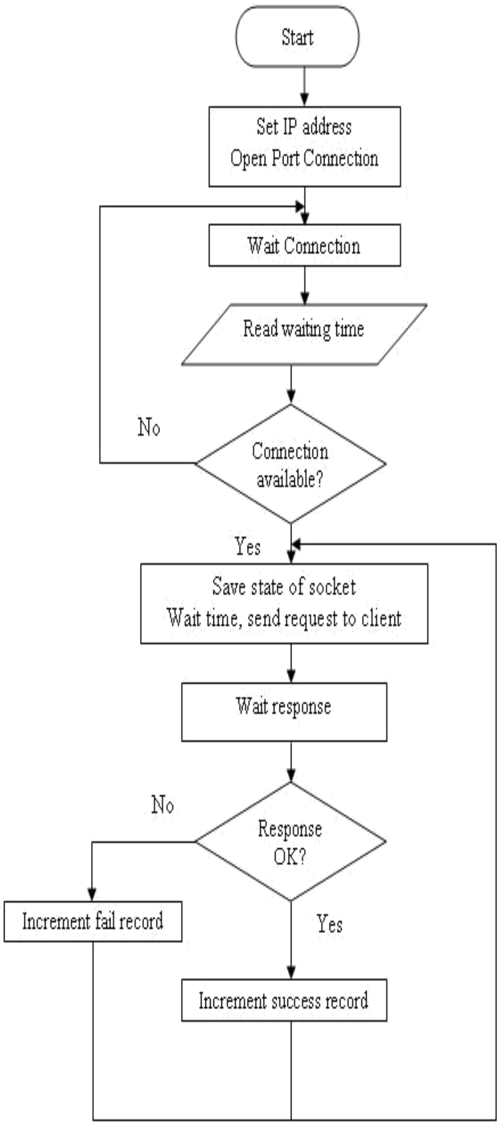

Figure 7 : Flow chart of Connectivity test program

C. Delay Test

In this test we measure the delay of packets in the GPRS link end to end for two direction upload and download. This method also uses two separate programs one on the client and one on the server. TCP protocol was implemented on both programs. The delay is computed during the time taken for the packet to traverse from sender to receiver.

D. Throughput Test

Throughput of a TCP transfer is calculated in the server end of a TCP transfer. The throughput is calculated by dividing the size in bytes of the transferred object with the time in seconds taken for the objects transfer. The transfer time is calculated from the arrival of the clients segment to the sending of the ACK to the clients segment. There is therefore some additional time in the transfer time. The additional time is typically close to a RTT, because the request is always small enough to fit in one segment. This metric gives us extra information to be used in the evaluation of stability of TCP transfers in a test.

V. RESULTS AND DISCUSSION

A. Round Trip Time (RTT) Performance

As mentioned earlier, the aim of this program is to send short packets from client to be received by the server application and finally received back at the client. It characterizes the end-to-end latency, which is important for time-critical applications and dynamic behavior of Internet protocols. Averages, maximum and minimum of round trip times were measured by this program from consecutive Ping commands. The test was repeated many times for several Ping packet sizes. Results are presented in fig. 8. Short packet's PING measurements are useful to characterize, for instance, the initial three-way TCP handshake. Two different locations in Tripoli were selected for stationary tests. These locations (Tripoli University and Alhani) were chosen due to their service availability.

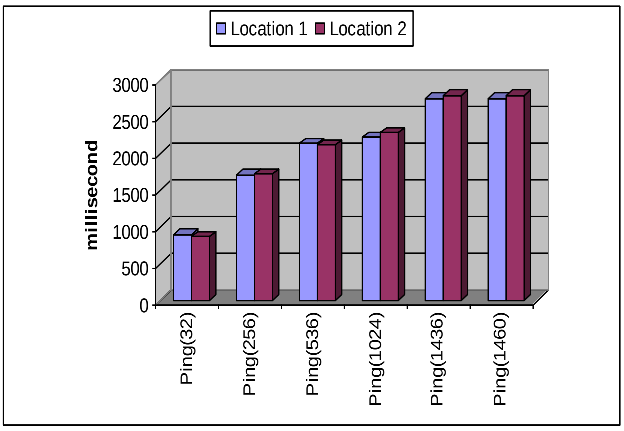

Figure 8: Average Round trip time measured with Ping command in two locations

Fig. 9 shows the comparison of round trip time between Libyana and Al-madar. Sending large size files (greater than 1Kilo bytes) over Al-madar network causes time out some times, due to the restriction in the router of Al-madar to prevent congestion caused by internet control message packet. However sending smaller file sizes over Al-madar gives better time response than Libyana which means traffic due colloquies in A-lmadar less than in Libyana network.

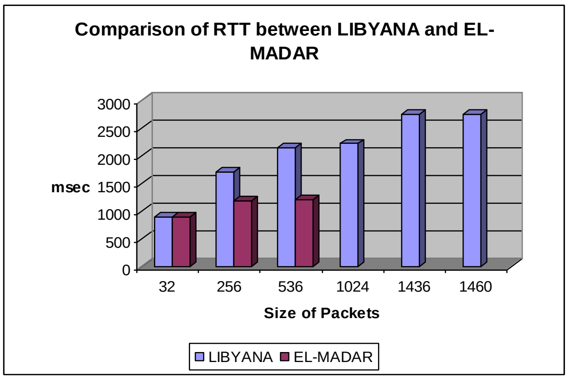

Figure 9: Comparison of RTT over Libyana and Al-madar

B. Impact of Latency on Service Performance

The RTT has effects on different mechanisms that directly impact end-user performance:

- Session setup delay. When a new service is activated, the mobile network (client) may first establish one or several Packet Data contexts in order to reserve resources.
- TCP performance. The establishments of a TCP connection and transmission rate are directly affected by the RTT.
- Service interactivity. Some services (such as voice or real-time gaming) that require small end-to-end delays may not be well supported over packet-switched technology if the round trip time is large. In general interactive response time should not exceed the well-known human factor limit of 100 to 200 milliseconds [5]. So a tradeoff should be found between efficiency and latency in the design of a sub-network.

C. TCP Performance

TCP performance impact in cellular networks is largely affected by the RTT. Large round trip delay makes initial data rates slow due to TCP long connection establishment. The amount of data a TCP can send at once is determined by the minimum value between the receiver's advertised window and the congestion window.

The receiver's advertised window is the most recently advertised receiver window and is based on the receiver buffering status and capabilities. The TCP sender also maintains a timeout timer for every packet sent. If no ACK is received after the expiration of this timer, the congestion window drops to one segment and the oldest unacknowledged packet is retransmitted.

D. UDP Performance

User Datagram Protocol (UDP) transport protocol is less problematic than TCP in wireless, as it does not require retransmissions and the protocol overhead is significantly lower. Some streaming services, such as Voice over IP (VoIP), use Real-Time Protocol (RTP) over UDP.

E. Throughput Performance

Data throughput is especially important in interactive data services, where the user expects to receive and send data files within a reasonable time. File downloads with different file sizes were performed with FTP application. The average throughput depends on file size due to TCP dynamic behavior. The throughput was measured during five file downloads of the same size. Table I and Table II shows the measured throughput during download and upload sessions respectively.

TABLE I: Application Throughput measured in download

File Size (KB)	**Download time (sec)**	**Throughput (Kbyte/sec)**
50	19.69	2.60
200	85.34	2.41
500	198.15	2.58

TABLE II: Application Throughput measured in upload

IP Receiving	41.208.168.105
Media Locator	ftp:// 41.208.168.105
File Size	200 KB
Download time	277.5 sec
Speed	0.74 Kbyte/sec

Table I shows the throughput measured in download, where the average throughput is about 20 Kbps for all size of files. In the other direction upload is about 5.9 Kbps. The difference is due to the fact that more time slots are assigned download than upload. The comparison of throughput between Libyana and Al-madar shows that, the throughput is better in Al-madar than Libyana as shown in Fig. 10. The reason is the number of customers in Al-madar is less than Libyana where GPRS signals is effected with number of calls.

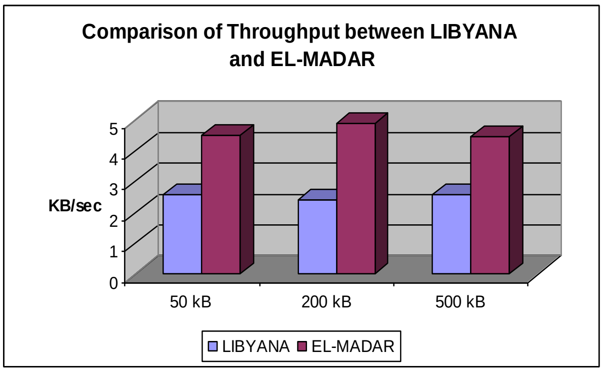

Figure 10: Throughputs Comparison between Libyana and Almadar

F. Effect of Connectivity

The aim of connectivity program is to measure the events and time that when the connection between GPRS client and server falls over one complete day. The result of this program show that, there is no disconnection happened during the whole day as shown in Fig. 11. This means the GPRS connection between client and remote server is always available where no traffic, but when burst traffic takes place the disconnection occur randomly.

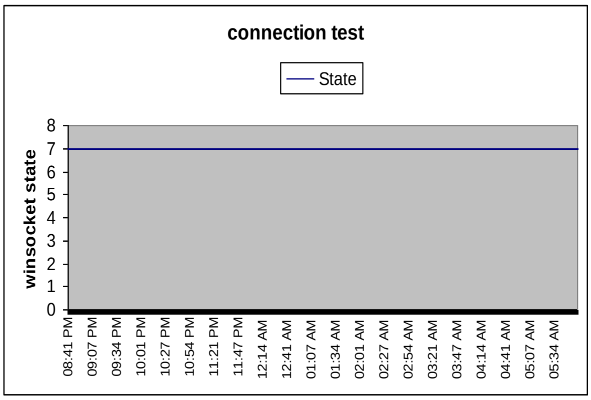

Figure 11: Connection availability over the whole day

G. Packet Delay Measurement

Fig. 12 shows the comparison results of delay time between Libyana and Al-madar for different size of files. The Al-madar GSM network has less delay time than Libyana GSM network for all type of files because Al-madar have traffic less than Libyana GSM network. When download small file (50 KB) few of seconds is different, while at large size of files (500 KB) the different is less than one hundred second.

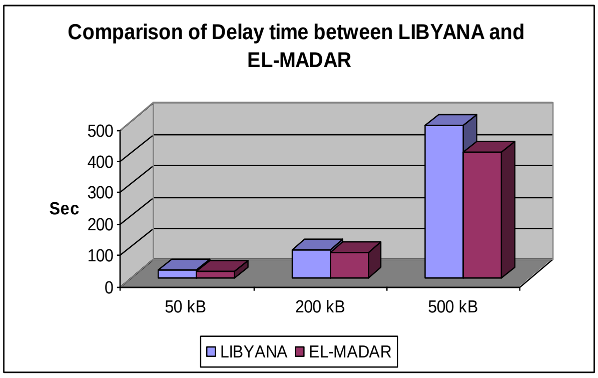

Figure 12: Comparison of delay time between Libyana and Al-madar

H. Packet Loss Measurement

In this experiment, the UDP protocol (connectionless oriented) is used as the base of all packet transmission over the network. Table III shows the percentage of packet loss and duplicate of packets during transmission from total packets for Libyana network.

TABLE III: lists the parameters of our Experiment

IP Receiving	41.208.168.105
File Size	50 KB
Format type	Microsoft Data Base Access(MDB)
Total No. packets sent	140
% Packet loss	10.1%-13.9%
% Packet Duplicate	7%-11%

VI. CONCLUSION

The objective of this paper was to specify and implement measurement system based on parameter of quality of service over GPRS networks. The protocols used for communication between the server and client are TCP and UDP. The implemented system was developed using Visual Basic package. The system was divided into number of sub-blocks. Each sub-block was studied separately using one of the quality of service parameters. The developed programs measured the GPRS system performance in varies conditions. The Ping Test was used for computing the round trip delay for wireless networks which is important factor in TCP protocol.

This work proved that the throughput is the most significant factor in determining usefulness of data download. Besides that, duplication of packets occur when the acknowledge packet does not arrive to transmitter, so it must allocate more packets in upload directions. Some TCP implementations are actually pretty good and getting close to the upper bound under different network conditions. The upper bound is the average throughput between a server and client, regardless of any latency.

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