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Establish a Methodology for Testing and Optimizing GPRS Performance Case Study: Libya GSM

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***Abstract*—**The main goal of this paper is to establish a methodology for testing and optimizing GPRS performance over Libya GSM network as well as to propose a suitable optimization technique to improve performance. Some measurements of download, upload, throughput, round-trip time, reliability, handover, security enhancement and packet loss over a GPRS access network were carried out. Measured values are compared to the theoretical values that could be calculated beforehand. This data should be processed and delivered by the server across the wireless network to the client. The client on the fly takes those pieces of the data and process immediately. Also, we illustrate the results by describing the main parameters that affect the quality of service. Finally, Libya's two mobile operators, Libyana Mobile Phone and Al-Madar al-Jadeed Company are selected as a case study to validate our methodology.

***Keywords*—**GPRS, performance, optimization, GSM

I. INTRODUCTION

MOBILE and personal communication is recognized as major driving force for industrial competitiveness and for sustained economic growth. On a personal level, telecommunications change the life styles of most people making them more mobile. For instance, as an amount of public or private transportation to work and leisure activities is increasing, people spend more and more time in transit. Time spend commuting is not work time or free time, but "mobile time". This has been recognized by developers of electronic technology which have made mobile electronics part of the mass consumer market.

Since the introducing of the Global System for Mobile communications (GSM) in 1992, mobile phones has become devices owned by more than half of the population in many countries. The market is expecting not only voice services but also more and more data services like Internet and multimedia. Principally, the intention is to bring together the GSM solution for wireless access with the Internet solution for information sharing, forming a solution for personal information access. These new solutions as well as demand for mass market mobile data services opening up for new business opportunities for operators and content providers.

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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. 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].

Wireless networks are problematic environments for data communications. 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.

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 [2]-[3].

Recently, the use of mobile data applications such as the GSM Short Message Service (SMS) has gained popularity. However, the GSM system can only support data services up to 9.6 kbit/s circuit switched.

GPRS, developed by the European Telecommunications Standards Institute (ETSI), is a packet switched data service for GSM that can allow bit rates, theoretically up to 171.2 Kbit/s per user.

However, commercial GPRS systems will be able to support rates up to 115 kbit/s. This paper presents current developments system that measures the parameter of Quality of Service (QoS) that effect on data transmission.

Therefore the goal of this paper is to develop software that can be used to investigate quality of service issues over Libya GSM networks as well as establish a methodology for testing and optimizing GPRS performance.

Libya has two GSM networks; Libyana Mobile Phone and Al-Madar al-Jadeed Company. The block diagram of the developed system discussed in this paper is shown in Fig. 1.

Fig. 1 Block diagram of developed system

II. GPRS MEASUREMENT

GPRS is an overlay to the circuit switched GSM network. The standardization of GPRS started in 1993 by the ETSI (European Telecommunication Standards Institute). The GPRS is a GSM phase 2+ services and is also an essential first step towards third generation mobile network (UMTS). GPRS increases the possible bandwidth for data transmission by introducing packet switching.

This section briefly discusses the measurements used to evaluate the performance of GPRS networks. GPRS measurements, which map into the model illustrated in Fig. 2, are divided into three categories: data performance, signal quality, and Radio Frequency RF performance.

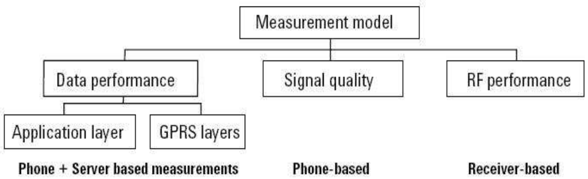

Fig. 2 GPRS measurement model

A. Radio Frequency (RF) Performance

This first category consists of network independent measurements such as interference, scanning, and spectrum analysis. The measurements require sophisticated RF test tools such as DSP-based RF measuring receivers.

B. Signal Quality

This category consists of physical layer measurements and a subset of RLC and MAC layer measurements. The measurements are made using a test mobile phone.

C. Data Performance

This category emphasizes data-transfer-quality measurements (as perceived by customers) and GPRS layer-specific measurements. Data performance measurements are used to establish quality benchmarks and to detail the performance of individual layers.

For our test purposes, one end node is the client (mobile) and the other is a measurement server. This 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. In the downlink the measurements are made by the same software that generated the uplink data.

D. Quality of Service (QoS) Parameters

Whatever parameters we measure, the ultimate objective at the application layer is to get the user perspective. Thus it is essential to benchmark performance broadly against certain standards.

ETSI GPRS recommendations define quality of service for users according to specific parameters, including throughput, and delay.

In this paper, our data performance measurements therefore focus on these parameters.

III. SYSTEM ARCHITECTURE

If you are using *Word,* use either the Microsoft Equation Editor or the *MathType* add-on (http://www.mathtype.com) for equations in your paper (Insert | Object | Create New | Microsoft Equation *or* MathType Equation). “Float over text” should *not* be selected.

The system is developed using client server architecture and it is designed to support unicast as mentioned before. During service system session the process of state transition with the server is illustrated as follows:

- *IDLE:* an initial state of the server where it waits for new clients connect request, after which it transits to WAIT state. When end request is received it comes back to IDLE from any other state.
- *WAIT:* the station that works as server waits for additional requests.
- *PLAY:* an active transmission state. The station is downloading or uploading depending on decision of the user.
- *STOP:* the station that works as server transit from play state to this state when transmission is halted.

A. Measurement Arrangements

For measurements we used a live GPRS network in Libya. The network implemented the 3GPP release 97. The GPRS network was able to support the number of time slots defined by the operator.

For end-to-end throughput measurements, we used software generating bulk data transfers over TCP. For measuring latency, we used a standard ping program. Measurements are useful for finding existing inefficiencies in data transport [4].

1. Test objectives

The objective of this test is to study the GPRS performance implications of those link characteristics that are typical for slow wireless links, and to study the effect of various QoS parameters on internet connection with respect to user.

We measure and analyze the effects of an unreliable link, which causes packet losses due to corruption, and a persistently reliable link, which cause excessive delays instead of packet losses, that represents in the internet community. We then employ various mechanisms to measure GPRS performance. It is clear that there is no single modification that gives an optimal result for all different test cases. However, we try to get a clear picture about the effects of different parameter that would work well in all test cases.

2. Test environment

We used two different computers to run these tests. One of the computers was used as end hosts and the other was running as a client on the GSM network.

Time 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 this test is done in real- time, it is important that the whole testing environment does not produce wrong results. Therefore, we dedicated these computers to our testing purpose only and there were no CPU consuming applications running, on them during this test.

B. Packet Drop Test

In these test cases we measure the packet drops during transmission over wireless network. First, we use UDP protocol at transmitter 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.

1. Packet drop Application programs

The test programs are implemented to send and receive these packets in a form of files, using UDP protocol and then present it to the user on the server by using socket method. There are two programs, one for sending packet data and other to receive it as following:

- Implement the client program, which can be used to transmit the data as packets.
- Implement the server program, which can be used to receive packet and store it in database.

2. Transmitting Packet

This program represents the most important stage, because it responsible for sending packets over the network. In this program, we transmit UDP packet by using tool called WinSocket installed in Microsoft Visual Basic package.

The UDP WinSocket is constructed by using WinSocket method (Network-Top_Down). To control the transmission, we call send and stop on the Winsockets, since we just used the UDP Winsocket to transmit the data.

3. Receiving Packets

To receive and present a data from a UDP session, a receiving program is implemented, which can use private method that describes the session.

C. Reliability Test

In this test we measure the number of times disconnection that may occurs in GPRS.

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.

D. 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.

1. Upload Experiment

In this experiment the (GPRS) client sends files with different sizes to server and computes the time taken to send the file

2. Download Experiment

In this experiment the (GPRS) client receives data files with different sizes of files from server and computes the time needed to receive each file.

E. Implement End-to-end Data Throughput and Latency Measurements

There are two basic end-to-end performance measurements that characterize the expected data service performance: round trip time and data throughput.

1. Implement Latency (Round Trip time)

The latency associated with the system must be introduced, as it is a key element when evaluating end-user performance. Although it is not considered as degradation by itself, it has a direct impact on upper layers behavior.

A common way to benchmark the network latency is to evaluate its round trip time (RTT). The network RTT can be defined as the time it takes to transmit one packet from e.g. a server to a terminal plus the time it takes for the corresponding packet to be sent back from the terminal to the server.

The RTT is a function of the packet size transmitted in downlink (DL) and uplink (UL). RTT (32 bytes, 32 bytes).This shows the RTT for Ping commands with small packets of 32 bytes [4]. It gives a good idea of the minimum network latency as well. Different packet sizes RTTs may be used to characterize various performance degradation effects.

RTT (32, 32) characterizes the minimum network latency and can be used to estimate the performance degradation associated with the application layer establishment protocols.

RTT (1500,32) [4] determines the latency associated with the transmission of large packets in UL and small packets in DL and can be used to estimate the impact TCP protocols dynamics (e.g. slow start) have on the end-user performance. It characterizes the end-to-end latency, which is important for time-critical applications (like gaming) and dynamic behavior of Internet protocols [6].

F. Throughput

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.

IV. Results and Discussions

In this section, we present and discuss the results from our application programs, and show their graphic user interface GUI. Furthermore, we investigate the quality of service parameters of GPRS.

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. Average, 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. 3. 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 (Ras-hassen and Wesait-bederi) were chosen due to their service availability.

1. 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. Usually large IP segments are recommended in order to minimize the effects of IP headers overhead. Although this is true, in slow systems where the latency can be high, large IP segments may take considerable time to be transmitted and introduce noticeable delays for the end user. For example, the latency when accessing a remote computer (e.g. using Telnet) should be kept small. In general interactive response time should not exceed the well-known human factor limit of 100 to 200 milliseconds [4]. So a tradeoff should be found between efficiency and latency in the design of a sub-network.

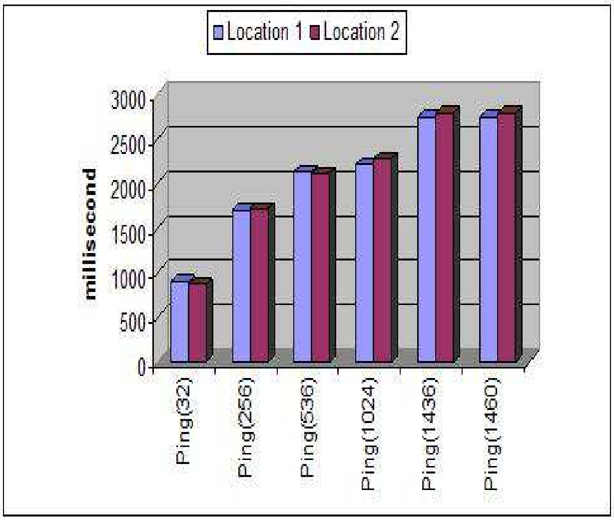

Fig. 3 Average Round trip time measured with Ping command in two locations

2. TCP Performance

Transport Control Protocol (TCP) [4] is currently used in 80% to 90% of data transactions in fixed networks. TCP provides reliable data transfer by means of a retransmission mechanism based on acknowledgments and retransmission timers.

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 [7]. 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.

3. 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 [8].

4. Comparison Round Trip time over Libyana and Al-madar

The Round trip time has been measured over Libyana [29] and Al-madar GSM networks [30], where the ping command has been sent over Libyana and Al-madar GSM networks.

Fig. 4 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.

B. 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. In our test throughput was measured during five file downloads of the same size. Table (I) and Table II show the measured throughput during download and upload sessions respectively.

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.

TABLE I
APPLICATION THROUGHPUT MEASURED IN DOWNLOAD

File Size	Download Time	Throughput
50 KB	19.69 sec	2.60 Kbyte/sec
200 KB	85.34 sec	2.41 Kbyte/sec
500 KB	198.15 sec	2.58 Kbyte/sec

The difference between them is due to the fact that more time slots are assigned download than upload. Further more the most traffic in GPRS are download.

1. Comparison Throughput over LIBYANA and AL-MADAR

The throughput has been measured over Libyana and Al-madar GSM networks by download different size of flies over Libyana and Al-madar GSM network. Fig. 5 shows the comparison of throughput between Libyana and Al-madar. From this we can see that the throughput is better in Al-madar than Libyana .

The reason is the number of customers in Al-madar is less than Libyana where GPRS signals is affected with number of calls.

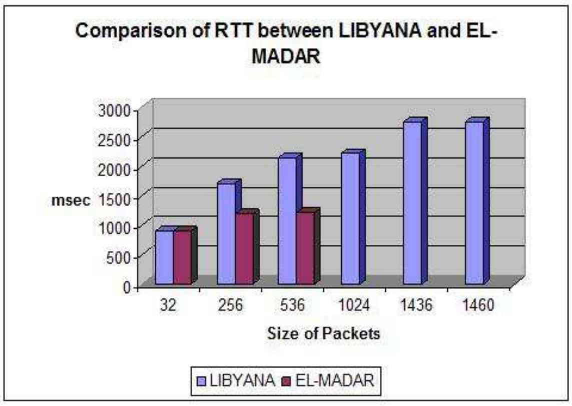

Fig. 4 Comparison of RTT over Libyana and Al-madar

C. Effect of Connectivity

The aim of this program is to measure the events and time that when the connection between GPRS client and server falls over one complete day.

The connectivity program capable of: Recording the state of socket.

- Sending messages from client to server at certain time.
- Saving all information in database.

The results of this program are shows in Fig. 6. From the figure we can see that no disconnection happened during the whole day which means that the GPRS connection between client and remote server is always available where no traffic, but when burst traffic takes place the disconnection occur randomly.

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

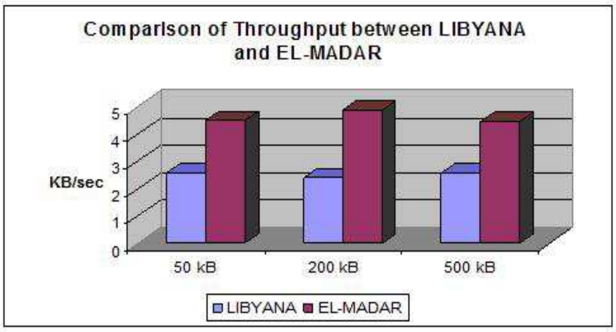

Fig. 5 Throughput in Libyana and Almadar

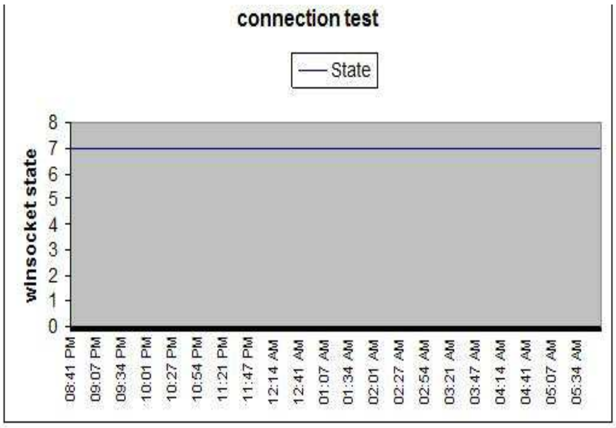

Fig. 6 Connection availability over the whole day

D. Details of the GUI of the Client/Server Application

The client/server application program that has been implemented in this project offers a user friendly graphical user interface that is organized in a main window with two main items.

The server (sender) runs continuously and waits for client (user at receiver side) to connect to it. The data to be send to clients by sender are text files.

1. Server Side Application

The server side application program is used for handling all requests from the clients for transmitting files over GPRS network. In the server program, the main window contains all data needed to compute the delay of file transfer.

2. Client Side Application

The client side application is a part of the main client/server application designed. It is concerned with the client connection to server side.

3. Packet Delay Measurement

One of the objectives of the above mentioned programs is to measure the time delay of packets sent from sender to receiver in upload and download sessions. Fig. 7 shows the results of time delay measured, while downloading a 50KB file from server.

The download operation is repeated over different times during the day. From Fig. 7 we can see that the maximum delay occurs at middle of the day where when usually there are lots of calls in the network especially arround 12:00 PM and 05:00 PM, bearing in mid that the signals of call have priority over GPRS signal.

The experiment has been repeated many times and end to end transmission delay has been measured. The average value has been measured by experiment and presented in Fig. 7.

4. Comparison Delay Time over LIBYANA and AL-MADAR:

The delay time has been measured on Libyana and Al-madar GSM networks, where different size of files over Libyana and Al-madar GSM networks downloaded by the above program. Fig. 8 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.

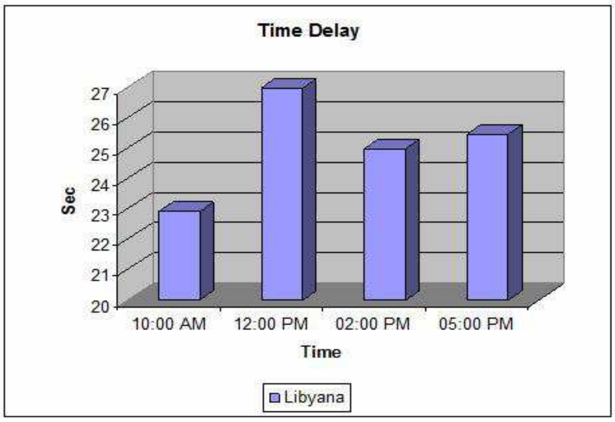

Fig. 7 Transmission time Delay (sec) for Data file over Libyana network

E. Packet Loss Measurement

The aim of this program is to send one hundred records from a database to the client computer. In this experiment we used UDP protocol (connectionless oriented) as the base of all packet transmission over the network. Table (III) lists the parameters of our Experiment.

F. Stability

If a throughput of one data transfer is considerably lower than another the stability of network is considered bad. Stability of transfers is important because in all of our test cases we have several concurrent data transfers.

Number of dropped packets affects directly the number of retransmissions and has remarkable effect on delay behavior.The number of dropped packets is calculated by the above mentioned application program.

The total number of drops in all repetitions could be used. Number of retransmission can be measured automatically with the number of drops. These delays can be seen in application, and may be followed by retransmission timeouts. Also lost acknowledgments and packets may cause an retransmission time out (RTO).

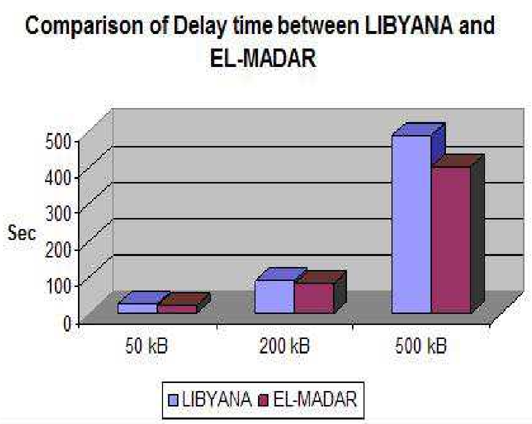

Fig. 8 shows the comparison of delay time between Libyana and Al-madar

TABLE III
THE PERCENTAGE OF PACKET LOSS AND DUPLICATE OF PACKETS DURING TRANSMISSION FROM TOTAL PACKETS FOR LIBYANA NETWORK

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%

G. Effect on security

The aim of this program is to send encrypted data over GPRS network to specific server by using RSA algorithm. The program is capable of:

- Generating Keys at the sender by using proper equation.
- Encrypting the file at the sender location.
- Sending the keys to the receiver before transmitting the file.
- Decrypting the file at the receiver using these keys.

V. CONCLUSIONS

Measuring the quality of service over GPRS networks is not new invention and the first measurement applications were made many years ago (GPRS-28). There are several commercial and non-commercial measurement software on the market.

The objective of this work was to specify and implement measurement system based on parameter of quality of service. The protocols used for communications between the server and client are TCP and UDP. In this work, we presented our measuring system together with experimental results that show that the system can be used on a network.

The implemented system in this project was developed using Visual Basic package. This system is divided into number of sub-blocks. Each sub-block was studied separately using one of the quality of service parameters.

The programs measured the GPRS system performance in varies conditions. In each program the user can change the system parameters, such as size of file, time of experiment.

The Ping Test developed in this project is an application program for testing the server connection. This application can be used for computing the round trip delay for wireless networks which is important factor in TCP protocol.

The developed system that is implemented in this project proved that the throughput is the most significant factor in determining usefulness of data download. Beside that, duplication of packets occur when the acknowledge packet does not arrive to transmitter, so it must allocate more packets in upload directions.

A good protocol should be able to deal with all different types of Internet congestion and should be close to the upper bound of the average throughput. 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.

We believe that handovers are a fundamental property of future mobile networking. We explored the effect of a change of networking characteristics triggered by handovers on end-to-end transport protocols using measurements of handovers between GPRS client and server, as well as experiment of handovers between GPRS client and server, we have shown that transfer of data has significant difficulties in adapting to new link characteristics after a handover. The adaptation time of the TCP rate to new link characteristics can take few seconds.

This paper presented and implemented end-to-end security mechanisms for the transferred data which is based on the RSA scheme.

All public keys used for encryption and decryption can be derived from the transmitter. To increase security level the key size of the RSA scheme is increased so that it may not be attacked by an intruder to compromise the transferred applications.

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