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Performance Evaluation of Round Trip Time in Libyan GSM network

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ABSTRACT
Due to the explosive use of Internet and wireless communication networks, Quality of Service parameters (QoS) has become a main research area in cellular wireless networks. One of the parameters of quality of service is delay in the cellular networks. In this paper we will focus on the Round Trip Time (RTT) delay, which is a particularly important variable for efficient End-to-End measurement delay in cellular networks. A test bench used is Libyana Network as a real time test tool for RTT performance measurement. End-to-end measurement methodologies of cellular networks have some issues that are not considered by traditional measurement techniques. Several experiments in different situations have been done in a real cellular network. In these final experiments, measurements of round-trip time over a Libyan GSM access network were carried out in Tripoli network. These measurements values are compared to the theoretical values. The packet round-trip time in the internal network (without including delays from the external network, i.e. Internet) is large (>600 ms), but its stable enough (small variation, and the packet losses are relatively rare.

Keywords; Performance, QoS, GSM, RTT, GPRS, PSTN.

I. INTRODUCTION

Today cellular networks beside a voice communications services also includes another services such as Email, SMS, audio, browsing of internet and video streaming. These types of services have strong constraints in performance of cellular networks and quality of services, to deal with these constraints; the RTT is used as measurement tools to analyze the end-to-end delay in GPRS networks. Using RTT as End-to-end measurements in GPRS networks have some particular problems that are not considered in the wired case [1],[2],[3]. For example, the Internet service provider (ISP) is bottle neck for all internet connection in Libya networks which are controlled the traffic for all internet communication, also gateway GPRS support node GGSN is considered as bottle neck in Libyan GSM network for the traffic from cellular network to ISP. The measurements presented in this paper are based on the end to end delay experienced by one GPRS user. The goals of this paper is to give service providers and GPRS network users in Libya a good performance indicator of GPRS networks in certain areas of Libya and to report on GPRS network user experiences [7]. Round Trip Time is dependent on a number of factors, including the data transfer rate of the source, the nature of the transmission medium, the distance between the source and the destination, the number of nodes between the source and the destination, the amount of traffic or bandwidth on the network that is being used, the number of other requests being processed by the receiver or nodes from end to end the transmission path , the processing capabilities of the source, receiver, and nodes, and the presence of interference [1].

This paper will focus on RTT because it is widely used in the measurement of performance sensitive applications [2], [8]. The measurements were taken in many places in the Tripoli city, one of them is Abusalem Suburb (wali Alahd Street). This area was chosen due to the fact that no wire line connections are available in this area. Therefore, GPRS networks are the main certain to extent the only reasonably priced data networks available in this area, which makes this study even more interesting. This is a unique situation in the State of Libya where almost all homes are connected to the internet through cellular networks. Furthermore, the situation in Abusalem Suburb with the unavailability of public switched telephone network (PSTN). Therefore, due to the choice of the area, where the measurements were taken, the results presented in this paper can be considered as a worst case scenario.

This paper is organized into eight sections including introduction, section II explores the related works, section III introduces the main objectives of the paper, section IV gives the measurement setup, section V deals with a round trip delay and the factors affecting it, section VI describes the experimental setup used to prove this relationship, in section VII results of practical measurements in real time hardware are presented, and the conclusion of this work is given in section VIII.

II. RELATED WORKS

Measuring the quality of service over GPRS networks is not new invention and the first measurement applications were made many years ago [7]. There is several commercial and non-commercial measurements software on the market [8]. Previous researches focused on simulations or active measurements in selected and usually well-controlled scenarios [9]. The choice had to consider the possible set of terminals, cell areas (urban, rural), and mobility patterns (stationary, highway), applications. The main research topics on RTT measurements on the Internet are: delay in a network. There are many suggested procedures in order to measure the delay of a wired link and wireless link.

Main techniques are based on sending variable size of packet. Each technique fits better depending on the characteristics of the path and the cross traffic. In general, all techniques work if there is not cross traffic. However, in a heavy loaded wireless network or in a path with cross traffic in many links, the delay in the measurement can be important [8]. Cross traffic is defined as packets which belong to other connections which interfere with probe packets.

The goal of RTT measurement is to measure one of the most important QoS parameters from end-to-end [10]. These works are related to the subject studied in this paper, because they contribute with methodologies and ideas to measure some end-to-end parameters. However, they do not solve the problems that arise in wireless networks. All these works are based on wired networks, where each link capacity is fixed and the packets share each link in a FIFO queue. This work builds a methodology that can be used for end-to-end performance measurement over cellular networks; where the previous assumptions are not necessarily true [10].

III. OBJECTIVES

The connection of internet via cellular networks in village or town is very difficult and sometimes inaccessible (during the bad weather) which has a high error rate. In the case of a failure it does not know at what level to find the failure, in order to prepare the mission, in addition, these villages are isolated; their operation is not monitored nor operates or has remote control. The main objective is to see how RTT (between source and destination) varies depending on the distance, baud rates, number of cells and packet length in a practical cellular network to operates in lower population density areas. Figure 1 shows the network measurement system.

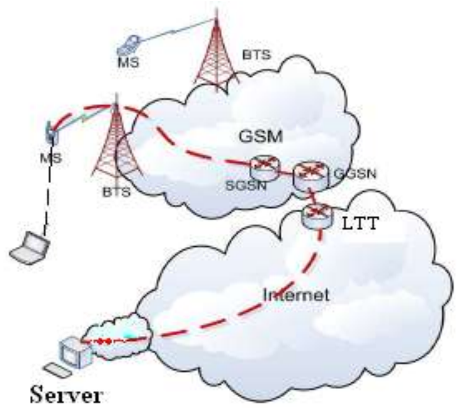

Fig.1: Network measurement System

IV. MEASUREMENT SETUP

The measurements were performed at Tripoli city in Abusalem Suburb (Wali Alahd Street) for GPRS of Libyan GSM network. A phone as GPRS terminal using Libyana SIM card were connected to a laptop using wireless connection link as shown in Figure (1). The terminal accesses the GPRS network over the radio interface. The base station controller (BSC) directs the packet data to the serving GPRS support node (SGSN), which authenticates and tracks the location of mobile stations. The SGSN forwards the user data to the Gateway GPRS support node

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(GGSN), which is a gateway to the external IP network [11].

In this test bed the delay was monitored by using ICMP, therefore only incoming packets were considered. A widely used communication link test tool is ping. It is a command line tool and is executed from the command prompt. Ping is a simple network test tool. It sends an Internet Control Message Protocol (ICMP) echo packet to a source. The destination host replies to the source address by echoing the same content that the source host sent. If the source host gets a reply from the destination host it knows the communication link between the source and destination works correctly at the IP layer [12]. It is possible to install a ping program on a smart phone. With this program it is possible to determine roundtrip time between a server on the Internet and the mobile device.

V. ROUND TRIP TIME (RTT)

Round-trip time delay (RTT), also recognized as round-trip delay (RTD), is the time required for a signal pulse or packet to travel from a precise source node (transmitter) thru path containing other node (receiver) and return again. The RTT can range from a few milliseconds (thousandths of a second) under ideal conditions between nearby spaced router nodes to several seconds under adverse conditions between router nodes separated by a large distance [12]. RTT almost always related to telecommunication, but may refer to the Internet, satellite radio router communications, and radar systems. RTT can point to a large category of transmissions, such as copper-cable Internet transmissions, wireless Internet transmissions, satellite transmissions, devices remote control transmission and cell phone transmissions.

In Internet wireless transmissions, the RTT may point to “ping time”, which is the amount of time data can be sent to a remote location and come back, and may be identified by using the ping command as shown in figure 2. In satellite transmissions, the RTT can be calculated by using the Jacobson/Karels algorithm [13].

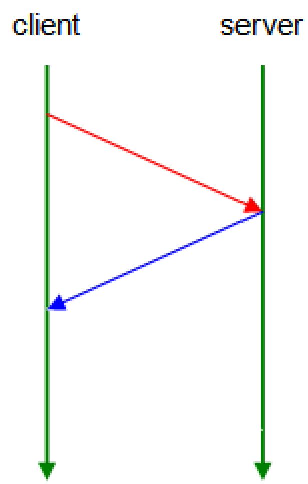

Figure 2: Round Trip Time

A. Round-trip time measurement

Round-Trip Time (RTT) is a measure of the time it takes for a packet to travel from a source across a network to a destination, and back. RTT is used by protocols like Transmission Control Protocol (TCP) to determine how long to wait for an acknowledgment before retransmitting [14]. In RTT measurement presented here, ICMP packets were sent using ping command to a server. Once the target server receives an ICMP packet, it sends it back to the sender immediately. From the time between sending and retrieval, the average RTT can be measured. The GPRS side round-trip delay between the computer that connected with mobile host and the server is determined by a number of factors. It depends on the available capacity in terms of the number of timeslots allocated to the mobile from GPRS, and whether GPRS is static time slots or dynamic.

GPRS of Libyana is static, where only one time slot are available. It has to be noted that in addition to GPRS data traffic volume the voice traffic has an impact on the GPRS quality of service. It is an important question whether daily fluctuations in GPRS service quality exist or not.

B. Factors affecting RTT time measurement

In a wireless network RTT is affected by several factors. One of them is latency, which is the time among a demand for data and the complete return of that data. RTT time depends on various factors including:

- Data transfer rate of the node.
- Nature of the transmission medium.
- Physical distance between the nodes.

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- Number of nodes in the RTT path.
- Number of other demands being manipulate by intermediate nodes.
- Speed between intermediate nodes and source node functions.
- Modulation mode.
- Presence of interference in the circuit.
- FEC (Forward Error Correction)

As stated above the round trip delay time is a function of various parameters of the wireless network and can be expressed by following equation:

$$\tau(t) = \sum_{i=1}^{m} \left(d_{l,i}(t) + d_{p,i}(t) + d_{q,i}(t)\right).$$

Let τ (t) denotes RTT at the time the ACK is received at the source node. Introduces $d_{l,i}(t)$ for the link delay of link i, $d_{p,i}(t)$ for the corresponding propagation delay, and $d_{q,i}(t)$ for the queue delay. Suppose the considered end-to end connection has m nodes.

C. Impact of Latency on Service Performance

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

- 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 [7]. So, a tradeoff should be found between efficiency and latency in the design of a sub-network.

VI. EXPERIMENTAL SETUP AND TESTING

The equipment's, including computer software used, are:

- Monitoring Software adapts to particular needs by providing tools for the supervision and control of the operations. This software can send different size of packets from client to the address of server application and finally received back at the client.

Figure 3 shows the user interface of the program written by Visual Basic Dot Net to perform the operations.

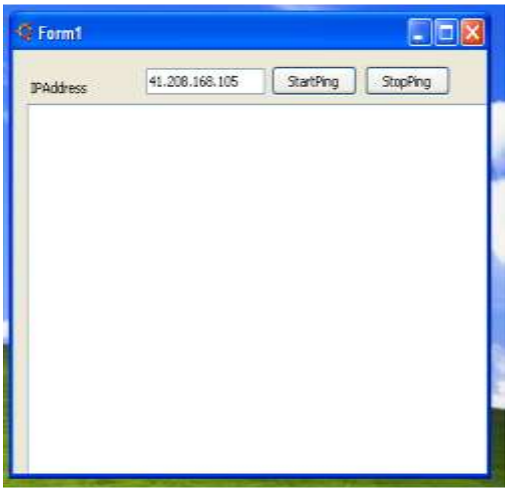

Figure 3: The user interface of RTT program.

The algorithms that implemented to perform this task are shown in figure 4.

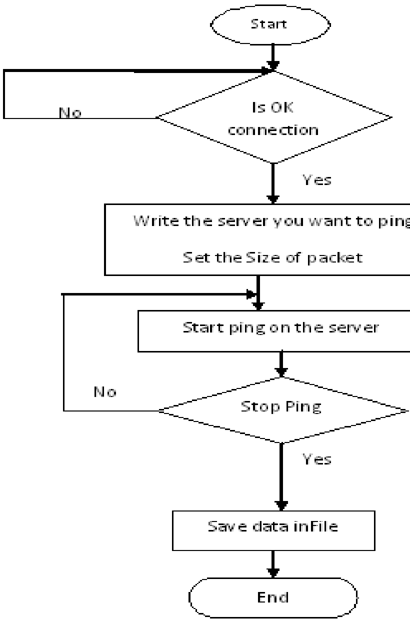

Figure 4: Flow chart of Program

A. Testing the communication path

There are several different reasons why a mobile service does not work. These can be divided into one of two broad classes:

1. There is something wrong at the application layer.
2. There is something wrong below the application.

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Viewed simply, this means that either something is wrong in the communication path between the two communicating devices, or something is wrong with the devices themselves. To distinguish between the two is not always a simple task, especially when the tester uses only a mobile telephone for testing the service. Trying to browse the Internet can give some hints as to where the problem resides. If the tester can connect to another WAP server and browse the Internet, then it will knows that the wireless communication link works. If the tester has a computer it is possible to use ping for testing for IP connectivity. Thus, it is possible to make sure the server "is alive". If the tester has a computer connected to the Internet via a mobile GPRS device it is possible to test the whole communication path. If the wireless link works with ping, but the service does not work over the same link, then there is a problem. A real challenge for the Verification is to find out who is responsible for a given problem. First they must verify that it is not the client or the server that is the source of the problem. As mentioned earlier it is generally the case that the faults are located at the endpoints of the communication chain. Secondly, after eliminating the client or the server as the source of the problem, these researchers must find the origin of the problem. This is difficult, because the researchers do not have access to the network of libyana system. In the following sections, different characteristics for different parts of the communication path are discussed.

B. Test of Mobile Phone

The mobile telephone is a very complex device and is a major part in the communication path. There is a web browser inside the telephone that must work according to the WAP and Internet specifications. All the protocol implementations must be accurate. There are a lot of different factors that must work in order for a mobile device to operate correctly. However, the problem could also reside at the server side. In order to connect to the GPRS network, the telephone must be configured. Parameter values including: Access Point Name (APN), proxy IP, and parameters for authentication must be configured in the telephone. When configuring a telephone, it is easy for the user to make a typing error when manually entering the parameter values. However, most operators can send the configuration to the mobile via SMS.

C. Test of Wireless link

In order for a service to work the mobile device must have a good radio connection. Thus loss of wireless communication is not an error when connectivity is temporary down, but rather that the user that does not currently have radio connectivity. For a normal GPRS user it is difficult to test the communication links inside the GPRS network. It is possible to set the APN, which determines which GGSN output interface the traffic should be routed through. However, most GPRS operators give their subscribers only one APN value for accessing the Internet. Therefore it is not possible to change this value in order to redirect the traffic within their GPRS network. There are many different elements that have to work in order to allow wireless communication to / from the Internet to work. If something goes wrong in the wireless link or in the internal GPRS networks, as a user can not easy to locate the error.

D. Test of external IP network (Internet)

An IP packet is forwarded though several IP network elements on its way to the server. All of these must work in order for the communication to work. For a communication path from a client to a server there might be a dozen or more IP routers that the packet must traverse. When testing IP networks, ping is excellent tool for testing the communication path. In most cases the path works for the services if it worked with ping. However, those problems were due to the WAP proxy, which should be regarded as a mobile specific network element. Once an IP packet has reached the public IP network, it is unlikely that some mobile specific traffic should get stuck while other passes through successfully.

VII. RESULTS OF PRACTICAL MEASUREMENTS

As mentioned earlier, the aim of this paper is to measure and evaluate the RTT time. 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 was measured from consecutive Ping commands. The test was repeated many times for several Ping packet sizes. Results are presented in tables. Short packet's PING measurements are useful to characterize, for instance, the initial three-way TCP handshake. The location (Sidi Salem- Wali Alahd Street) was chosen due to their service availability.

Table 1 to table 4 show the RTT for selected place which is Abusalem Suburb (Wali Alahd Street). The scenarios are try to ping with standard size of protocol ICMP on knowing server outside Libya with an URL address. We repeat this experiment many times and in different time.

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Table 1: RTT www.google.com.ly (packet 32 Byte)

No	Time	Round Trip Time(ms)			Packet loss (%)
		Max.	Avg.	Min.	
1	09:00	795	746	678	0
2	09:30	810	781	735	0
3	10:00	987	811	729	0
4	10:30	781	754	684	0
5	11:00	1011	846	742	0
6	11:30	954	827	751	25%
7	12:00	878	824	739	0
8	12:30	814	778	701	0
9	13:00	1107	844	738	25%
10	14:00	890	748	707	0
11	15:00	862	819	798	0
12	16:00	1054	818	907	25%
13	17:00	1098	844	829	25%
14	18:00	978	827	809	0
15	19:00	937	814	799	25%
16	20:00	889	839	829	
17	21:00	909	814	784	25%
RTT minimum (total) = 678 ms RTT Maximum (total) = 1098 ms					

Table 2 shows the RTT for selected place with packet size of 256 Byte in different time for the two days.

Table 2: RTT www.google.com.ly (packet 256 Byte)

Time		No. of packets sent	Round Trip Time(ms)			Packet loss (%)
Date	Time of exp.		Max	Avg	Min	
20 Dec.	16:00	25	1278	1239	1181	25%
	17:00	50	1315	1286	1218	30%
	17:45	50	1295	1269	1247	25%
	18:30	50	1301	1283	1264	25%
21 Dec.	20:30	10	1327	1271	1258	11%

Table 3 shows the RTT for selected place with packet size of 1024 Byte in different time for the two days.

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Table 3: RTT www.google.com.ly (packet 1024 Byte)

Time		No. of packets sent	Round Trip Time(ms)			Packet loss (%)
Date	Time of exp.		Max	Avg	Min	
27 Dec	16:00	25	3580	1838	1036	87 %
	17:00	46	2768	1901	1157	90 %
28 Dec	15:00	19	2962	1734	1072	84 %
	19:00	37	3165	1684	1105	89 %

Table 4 shows the RTT for selected place with packet size of 32 Byte in different time for the two days to server inside in Libya.

Table 4: RTT www.Libyana.net.ly (packet 32 Byte)

Time		No. of packets send	Round Trip Time(ms)			Packet loss (%)
Date	Time of exp.		Max	Avg	Min	
1 Jan	14:00	25	862	756	688	25%
	16:00	25	914	781	673	15%
2 Jan	15:00	30	824	727	662	25%
	19:00	30	851	741	673	30%

Table 5 to table 7 show the RTT for different places with GPS location for every experiment. The scenarios are ping with standard size of protocol ICMP.

Table 5: RTT www.google.com (packet 32 Byte)

Position (32°51′31″ N 13° 05′ 05″ E) location Gargaresh							
Time		No. of packets sent	Size of packet (Byte)	Round Trip Time(ms)			Packet loss(%)
Date	Time of exp.			Minimum	Maximum	Average	
21/5/2021	16:00	21	32	190	3479	1136	35 %
		24	256	293	2988	705	11 %
		19	512	413	2733	846	21 %

		21	1024	412	2365	892	27 %

Table 6: RTT www.google.com (packet 32 Byte)

Position (32°51´34″N 13° 13´17″ E) location Tripoli university gate							
Time		No. of packets sent	Size of packet (Byte)	Round Trip Time(ms)			Packet loss (%)
Date	Time			Min	Max	Ave	
21/5/2021	17:00	23	32	160	1586	424	15 %
		20	256	313	1883	1021	11 %
		21	512	455	1382	869	19 %
		21	1024	832	2891	1740	22 %

Table 7: RTT www.google.com (packet 32 Byte)

Position (32°51´28″N 13° 21´44″ E) location gate of Tajora							
Time		No. of packets sent	Size of packet (Byte)	Round Trip Time(ms)			Packet loss (%)
Date	Time			Minimum	Maximum	Average	
21/5/2021	19:00	22	32	158	3479	1359	0%
		23	256	201	2213	777	4%
		34	512	407	1537	767	9%
		23	1024	463	3303	1662	12 %

VIII. CONCLUSIONS

There is no easy way to expose such problems, especially without access to the network element logs and management system for each operator's network. Even though these kinds of problems are difficult to detect. These problems do not happen very frequently. Ping is useful for testing the communication path at the IP level. These tools can be helpful, especially if the problem resides within the public IP network (Internet). If an error is located within the GPRS network, these tools can only indicate to the tester, that the problem resides somewhere within the GPRS network. If the communication path is unstable it is difficult to troubleshoot. When the problems only appear randomly, it is difficult to recreate the failure. The most common reason for an unstable path is due to the wireless link. If the radio connectivity is weak, a little distortion can interrupt the communication between client and server. This work proved that the RTT is the most significant factor in determining usefulness of data transferred during cellular networks. Besides that, high values of RTT means that the cellular network is not accommodate the requirement of application users.

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