Portable IP-Based Communication System using Raspberry Pi as Exchange

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***Abstract*—The importance of communications and information trigger the development of communication network technology in order to be easy, cheap, efficient and has wide coverage area. This research designs an easy, cheap, and efficient IP-based communication system for voice and video. The aim of this research is to know the performance of telecommunication system for voice and video communication in terms of throughput, jitter, delay, and packet loss according to ITU-G.114 standard. This research also optimizes coverage area by adding 8 dBi omni-directional antenna. The arrangement of communication system is done by Raspberry Pi as server using ArteriskFreePBX software. Linphone and Bria as softphone in client devices. Then configure the SIP account to make voice and video communication. The test scenarios are communication between laptops, communication from laptop to smartphone, and communication between smartphones. The results obtained that the farthest distance from the wireless adapter between the server and the clients is 100 m. The testing network performance parameters based on ITU-G.114 standard, the farthest distance that can communicate between Smartphones is at 140 m for audio and 80 m for video, Laptop to Smartphone at a distance of 180 m for audio, 80 m for video. Communication between Laptop for video communication at a distance of 120 m and for audio can communicate well up to 200 m distance. This research is expected to be used as reference and alternative a cheap and portable telecommunication device.**

Keywords—IP-based; Raspberry Pi; Client-server; Throughput; Jitter; Packet Loss; Delay

I. INTRODUCTION

Telecommunications technology is growing over time, triggering the emergence of various technologies such as IP based voice communications network technology (VoIP). VoIP works by converting voice into digital packets and sending them over IP networks. At the beginning of its development, VoIP can only be used between PCs with low quality. Currently, the technology allows multimedia communication between PC to telecommunication device or communication between the telephones with good quality. Therefore, VoIP service has been widely used.

Along with the increasing mobility of users and the need for information and communication that has penetrated into almost all society, telecommunication equipment is expected to be easy, cheap, efficient, and has wide coverage area. Increasing the need for a cheap and easy communication system, requires devices that are not only cheap but also easy to use while moving. There have been many cheap and small electronic devices that can be utilized as a telecommunication network equipment system. Whether it is a telecommunication terminal or as an exchange of information systems. CubieBoard [1], Raspberry Pi [2], Odroid [3], HummingBoard [4], or Arduino [5] are various mini computers that have the ability to be used as exchange of telecommunication systems.

Some research on IP-based network technology using Raspberry Pi has been done. Research in [6] uses Raspberry Pi as a server for voice communications using G.711 codecs that can simultaneously call up to 12 calls with CPU usage below 70%. Research [7] performs voice communication testing along with server and client configuration. Likewise research [8] uses Raspberry pi for streaming video. These studies prove that mini computers such as Raspberry Pi can be used as server for telecommunication systems. However, the researchers tested only for voice call or video streaming whereas mini computers are believed to have capabilities that can be used for voice and video in a telecommunication system.

Other studies that use Raspberry Pi have been done to show the ability of Raspberry Pi. This can be seen in wireless sensor network applications that use raspberry pi as nodes or as servers [9], [10], [11], smart home application for home controlling [12], [13], [14], and health care monitoring system [15], [16], [17]. These studies prove that raspberry Pi can be used to process various applications even can be used as a back end of a network system. Therefore, this research designs an IP based communication system which provides voice and video communication by utilizing Raspberry Pi as an exchange. The purpose of this research is to know performance of Raspberry Pi as an exchange for communication in terms of throughput, jitter, packet loss, and delay. This study also optimized coverage area of WiFi used by adding omni-directional antenna with gain of 8 dBi. This research is expected to be used as reference and alternative telecommunication device.

II. LITERATURE RIVIEW

VoIP is a technology that can be used for voice communications by using data communication lines on a network. This technology works by converting voice into digital code and streamed over a network that transmits voice packets over an IP protocol network. In other words this technology is capable of transmitting voice traffic in the form of packets over the IP network. In general the VoIP diagram can be seen in Fig. 1.

A. VoIP Network Supporting Protocols

- Model Open System Interconnection (OSI) Layer

  OSI Layer is an abstract and layered description for communication and design of computer network protocol. The OSI model was developed by ISO (International Organization for Standarization) in 1977. The OSI model consists of seven layers: Application, Presentation, Session, Transport, Network, Data Link, and Physical [18].

- Transmission Control Protocol/Internet Protocol (TCP/IP)

  TCP/IP is a data communications standard used by the Internet community in the process of exchanging data from one terminal to another terminal within the network. The goal is to connect hosts on different networks, or may be geographically dispersed over a large area. [18].

- User Datagram Protocol (UDP)

  UDP is one of the main protocols above IP which is a simpler transport protocol compared to TCP. UDP is used for situations that do not emphasize reliable mechanisms. The header of UDP contains only four fields: source port, destination port, length and UDP checksum where the function is almost the same as TCP, but the checksum feature in UDP is optional [18].

- Protocol H.323

  H.323 is a standard to defines the protocol components. The protocol also provides multimedia communications services that communicate audio, video and data on time through packet-based networks. Packet-based networks include IP, Internet Packet eXchange (IPX), Local Area Network (LAN), Enterprise Network (EN), Metropolitan Area Network (MAN), and Wide Area Network (WAN) [19].

- Session Initiation Protocol (SIP)

  SIP protocol is published by IETF (RFC 3261) after VoIP. SIP is a signaling protocol on the application layer that works to build, modify and terminate a multimedia session involving one or more users. A multimedia session is the exchange of data between users that can include voice, video and text [19].

B. Video Over Internet Protocol (VoIP)

VoIP is one of the multimedia applications that allows communication for data, voice, and images in a real time. The form of this application is the communication through the communication media for video and audio between users directly [7].

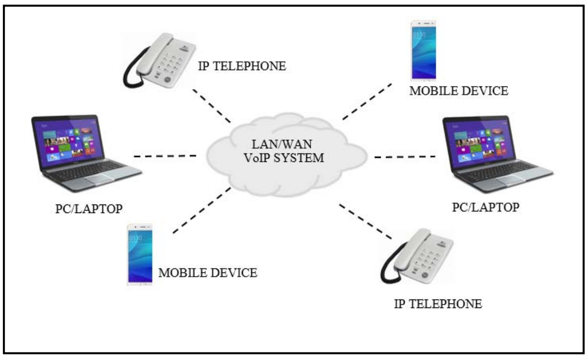

Fig. 1. General VoIP System

C. Network Performance Parameters

There are many possibilities may occur when sending data from source to destination that leads to a decrease in the quality of communication. Therefore, it is necessary to measure parameters to determine the quality of services of a network. These parameters include throughput, jitter, packet loss, and delay [20]:

- Delay, an accumulation of delay times from source to destination. The recommendation standard for delay by ITU-G114 can be seen in Table 1.

TABLE I. ITU-G.114 RECOMMENDATION STANDARD FOR DELAY

Delay **(ms)**	**Quality**
0 – 150	Good
150 – 400	Fair
> 400	Poor

- Jitter is the difference between the arrival time intervals between packages in the destination terminal. Variations in delay time may be caused by congestion, lack of network capacity, packet size variation, and packet disruption. Table 2 shows the recommendations of jitter values by ITU-G.114.

TABLE II. ITU-G.114 RECOMMENDATION STANDARD FOR JITTER

Jitter **(ms)**	**Quality**
0 – 20	Good
20 – 50	Fair
> 50	Poor

- Throughput is the rate of effective data transfer, measured in bits per second (bps) or bytes per seconds (Bps). The throughput represents the total number of successful arrival packets observed on the destination side over a specified time interval divided by the duration of the time interval.

- Packet Loss is the number of packets lost during packet data transmission in the network. Some causes of packet loss is the presence of noise, collision and congestion caused by the occurrence of excessive queue in the network. Packet Loss is still acceptable according to ITU-G.114 standard if the number of missing packet levels ranges from 0 - 0.5% shown in Table 3.

TABLE III. ITU-G.114 RECOMMENDATION STANDARD FOR PACKET LOSS

Packet Loss **(%)**	**Quality**
0 – 1	Good
1 – 5	Fair
> 10	Poor

D. Asterisk FreePBX

Asterisk is an open source that gives a real-time connectivity for VOIP networks. Asterisk can be organized on the core of an IP or on a hybrid PBX, call switching, router management, enable various features and connect callers with IPs in the outside world with analog and digital networks. Asterisk can run on a variety of operating systems including Linux, Mac OS X, OpenBSD, FreeBSD and Sun Solaris that provide all the features that PBX needs [21]. FreePBX is an application with Graphical User Interface (GUI) that can control and manage Asterisk. The features supported by FreePBX include Voicemail, Ring Group, Follow Me, Music on Hold, Call Queues and others [22].

E. Raspberry Pi

Raspberry Pi is a mini single-board computer with size of a credit card, developed by the Raspberry Pi Foundation. Raspberry Pi can be used in electronic projects and many things that desktop PCs usually do that support Linux-based operating systems. [5]. The Raspberry Pi board contains a processor and graphics chip, RAM, and various interfaces for external devices. It is cheap, powerful, and it does not consume a lot of power [7]. Physically, Raspberry Pi shows in Fig.2.

F. Softphone

Softphone is a software that acts like a phone device. The software is similar to a regular telephone machine, with the usual buttons on the phone. Softphones can be connected to internet phone service providers, such as Skype, and can also be used to connect to an IP PBX network on a LAN [6].

III. SYSTEM DESIGN

A. System Overview

This chapter discusses the design and installation of the hardware and software of the system design. The system is divided into two main parts: installation and configuration. Installation and configuration is done both on the server and on the client. For more details, block diagram of design can be seen in Fig. 3.

Performance evaluation is done by using star topology where raspberry pi as server and clients using laptop and smartphone. The network configuration is shown in Fig. 4.

Fig. 2. Raspberry Pi

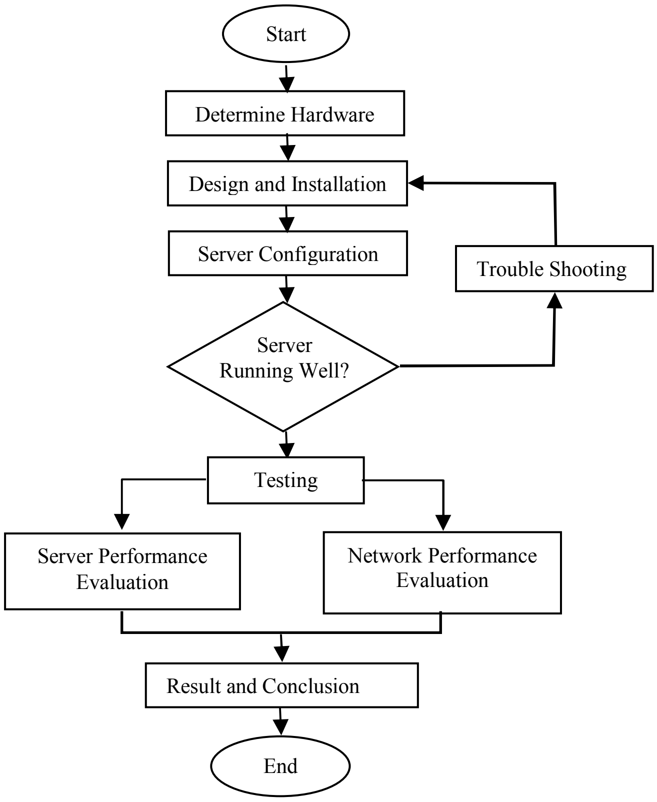

Fig. 3 Diagram Blok of System Design

B. Device Specifications

Various hardware and software are used in system design.

1) *Hardware*
- Raspberry Pi
- Memory Micro SD Sandisk 16 GB
- USB Wireless Adapter TP-Link WN722N
- Omni-directional Antenna 8 dBi TL-ANT2408CL
- Laptop
- Android Smartphone

2) *Software*
- Asterisk FreePBX (RasPBX)

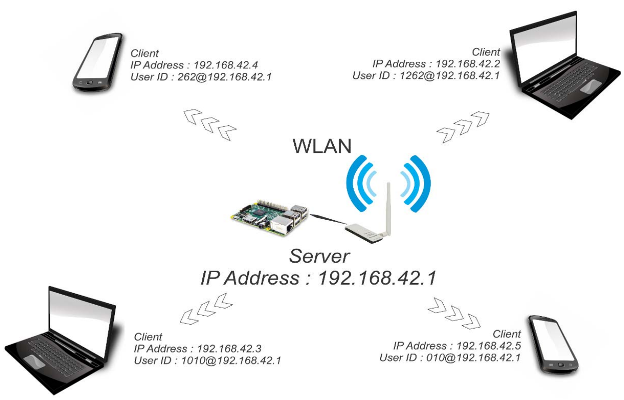

Fig. 4. Network Topology

- Softphone: Linphone (PC) and Bria (Smartphone)
- PuTTY (Raspberry Pi Remote Desktop)
- Web Browser (Web Server Configuration)
- Wireshark (Measuring QoS Parameters)
- Wifi Network Monitoring / Analyzer

C. *Server*

The activity on the server consists of three main processes: Asterisk FreePBX installation, testing network connections, and adding SIP accounts. Fig. 5 illustrates the main processes on the server.

D. *Client*

There are several processes to the clients. The first is to configure the IP address. Then, install the softphone application. The last is to configure the account on softphone. The client configuration diagram can be seen in Fig 6.

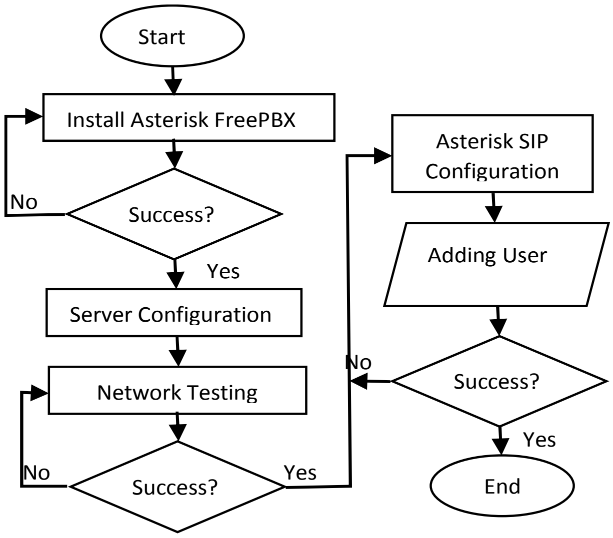

Fig. 5 Server Process

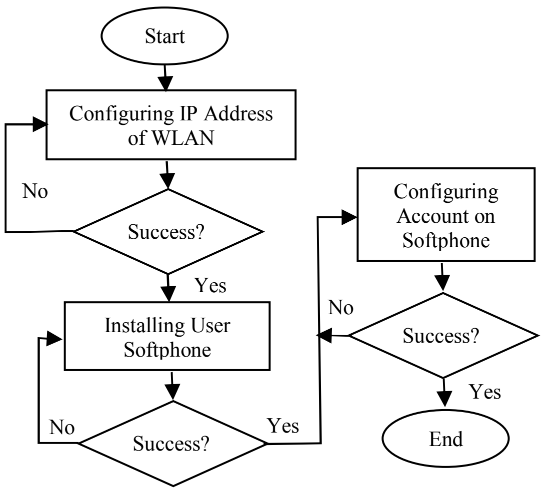

Fig. 6 Client Process

IV. Testing and System Analysis

The system test is done by three communication scenarios: Laptop to Laptop, Laptop to Smartphone, and Smartphone to Smartphone as shown in Fig. 7. The system uses Laptop Asus A46CB series with wireless card adapter "Qualcomm Atheros AR9485WB-EG" while Smartphone devices are Samsung Galaxy S3 and Asus Zenfone 2.

A. *WLAN Testing (WiFi Adapter)*

Testing for signal strength of WLAN conducted at line of sight condition using WiFi Network Monitoring application. Testing is done by varying distance between server and client from 10 meter to 100 meter. Raspberry Pi is equipped with WiFi adapter Tp-Link TL-WN722N with 8 dBi omni-directional antenna. The measurement results are shown in Table 4.

TABLE IV. Measurement of Signal Strength

Distance (meter)	Signal Strength (dBm)	
	Laptop	**Smartphone**
10	-48	-52
20	-52	-60
30	-54	-66
40	-58	-70
50	-60	-73
60	-63	-76
70	-69	-82
80	-71	-85
90	-74	-89
100	-82	-91

B. *Network Performance*

The measured network performance parameters are delay, jitter, throughput, and packet loss. Measurements are made by making calls with three types of communication scenarios. The three scenarios are:

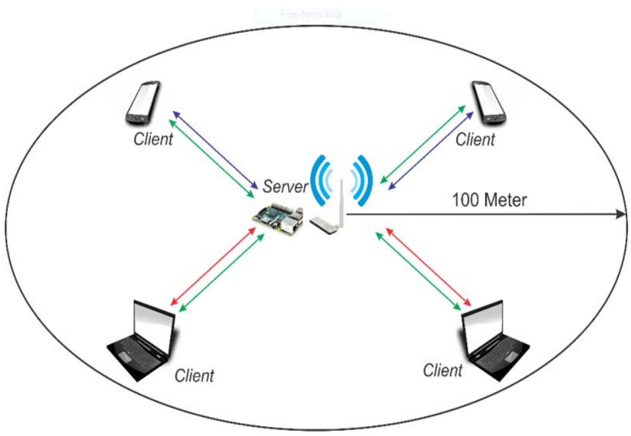

Fig. 7 Network Topology

1) Laptop to Laptop Communication: Using 2 Laptops Asus with Qualcomm Atheros AR9485WB-EG wireless adapter.
2) Laptop to Smartphone Communication: Using Laptop Asus and Smartphone Samsung Galaxy S3.
3) Communication Smartphone to Smartphone: Using Smartphone Asus Zenfone 2 and Smartphone Samsung Galaxy S3.

The following are the results of measurement of network performance parameters for audio and video communications.

- Throughput

Fig. 8 and Fig. 9 show the average throughput values for both audio and video communications. They can be seen that communication can be done up to a distance of 200 meters. At a distance of 200 meters, PC to PC communications has the number of average throughput about 75 Kbps for audio and 325 Kbps for video communication. The communication between smartphones has the smallest throughput value for audio and video communications that are approximately 45Kbps and 200 Kbps respectively.

- Jitter

Jitter measurement results for audio and video communications can be seen in Fig. 10 and Fig. 11. Theoretically, the jitter is greatly influenced by the distance which will result in the variations on the arrival time interval between packages to the destination. Based on the data shown in Figure 9, voice communications up to 200 meters for message between PC and from PC to Smartphone are included into good category as per ITU-G.114 standards. The communication between Smartphone can communicate well up to a distance of approximately 165 m.

The measurement of jitter for video communication is shown in Fig. 11. The figure shows that Laptop to Laptop communications has a good quality up to 200 meters. Communication Laptop to Smartphone can still communicate well to a distance of 140 m, while Smartphone to Smartphone communication up to a distance of 120 m has a jitter value of 43 ms.

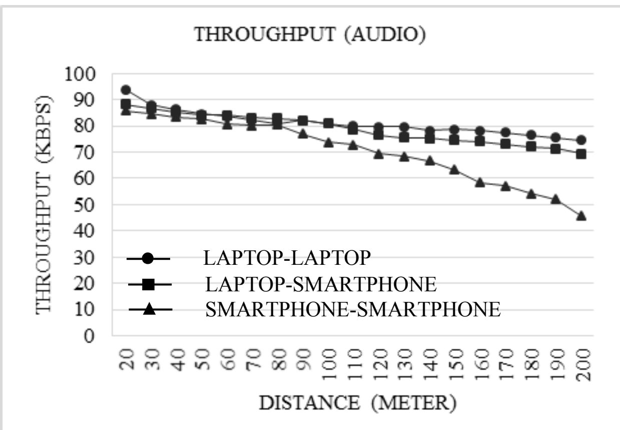

Fig. 8 Average throughput for audio

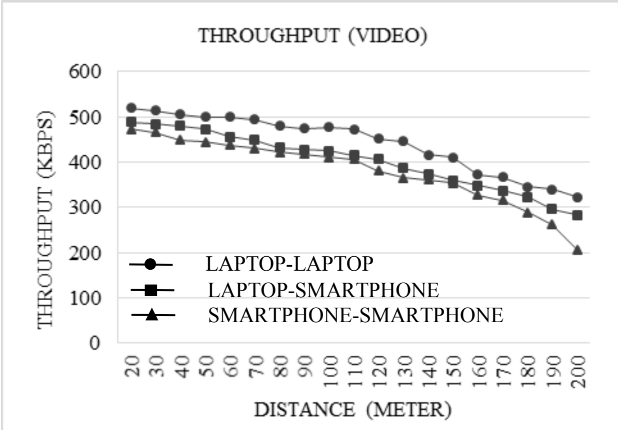

Fig. 9 Average throughput for video

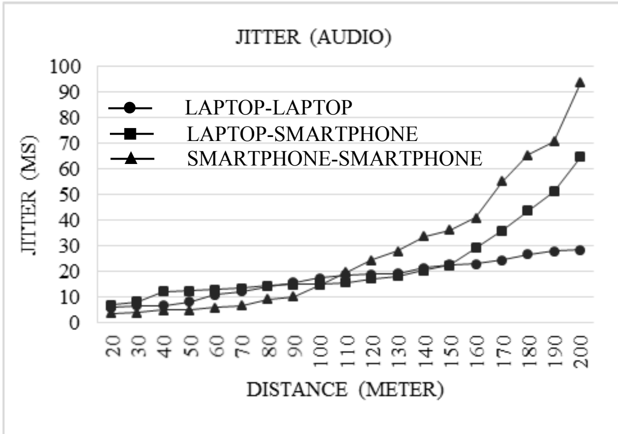

Fig. 10 Jitter for audio

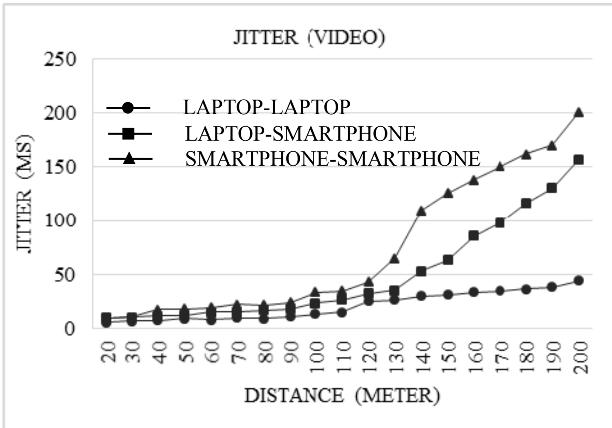

Fig. 11 Jitter for video

- Packet Loss

Packet loss is a very important parameter in data communication because a data packet contains an information. The greater the packet loss in the data transmission system the worse the information received and may even be lost.

Fig. 12 shows the percentage of loss packets for audio communication. It can be seen in the figure that the longer distance between the sender and the receiver the higher the packet loss. Based on the standard loss packet of ITU G.114, Smartphone to Smartphone communications can only be used up to a distance of 140 meters with a percentage about 2.86%. Laptop to Smartphone communication up to 180 meters with percentage 3.8%. Communication between Laptop can still communicate well up to a distance of 200 meters.

Fig. 13 shows the percentage of packet loss versus distance for video communications. As can be seen in the picture that the distance can affect the quality of the network, the longer the distance of communication the greater the loss package. Based on ITU-G.114 standard, Laptop to Laptop communications can only be used up to a distance of 120 meters with the percentage of packet loss about 4.96%. Communication Laptop to Smartphone and Smartphone to Smartphone can communicate well at a distance of 80 meters with percentage of packet loss 3.72% and 4.88% respectively.

- Delay

Fig. 14 and Fig. 15 show the delay values for voice and video communications. The pictures show that both laptop to laptop or laptop to smartphone communication can still be done well up to 200 meters both audio and video. Laptop to laptop communication at a distance of 200 m shows a delay value about 35 ms for audio while for video approximately 50 ms. Laptop to smartphone communication has a delay value at a distance of 200 m about 75 ms and 160 ms for audio and video respectively.

V. CONCLUSION

1) Raspberry Pi is equipped with WiFi adapter and 8 dBi omni-directional antenna has coverage area up to radius of 100 meters with -82 dBm for Laptop and -91 dBm for Smartphone.
2) The average throughput for the three types of communication scenarios with a communication distance of 200 meters lies in the range from 40 to 100 kbps for audio communications whereas for video communication it has a throughput between 200 and 500 kbps.

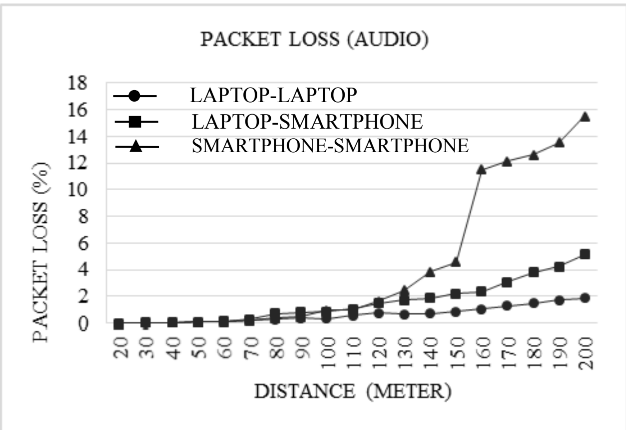

Fig. 12 Packet loss for audio

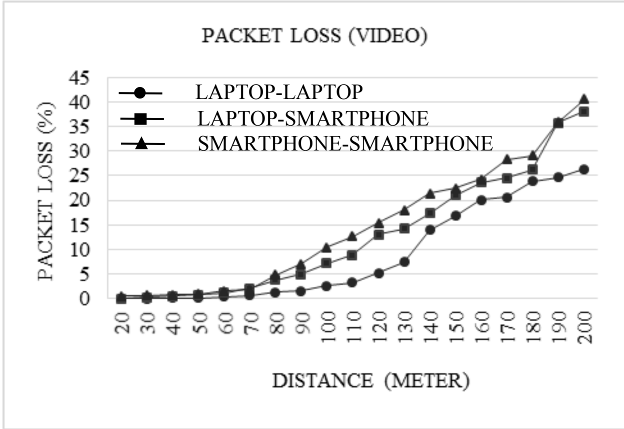

Fig. 13 Packet loss for video

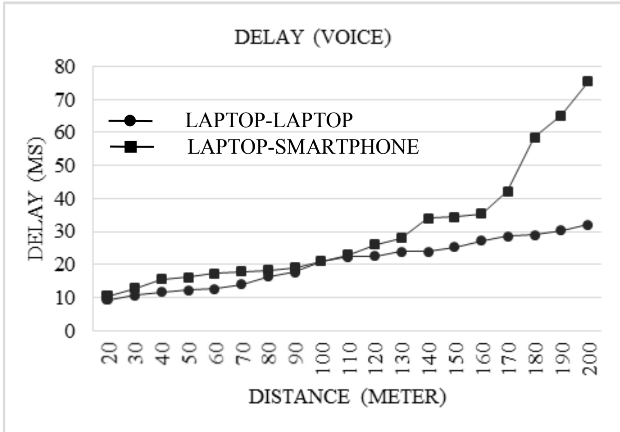

Fig. 14 Delay for audio

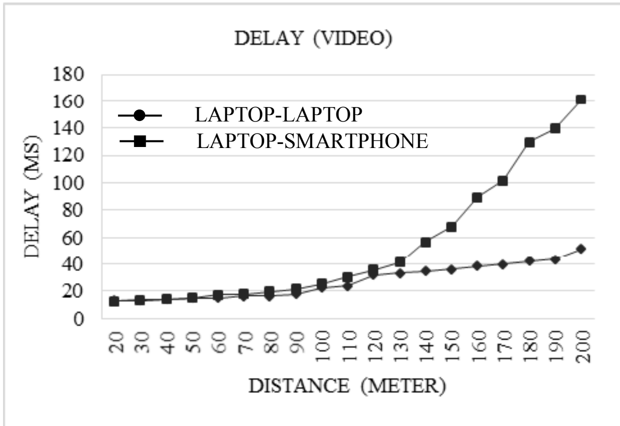

Fig. 15 Delay for video

3) Based on ITU-G.114 Standard for jitter:
   - Communication from laptop to laptop for audio and video communications can be done up to a distance of 200 meters.

- Communication laptop to Smartphone for audio communication can be done up to 200 meters while for video communication only up to 140 meters
- Communication between Smartphone for voice communications can be done up to a distance of 160 meters while for video communication only up to a distance of 120 meters.

4) Based on the ITU-G.114 standard on packet loss:
   - Communication between laptops for voice communication up to 200 meters and for video communication up to 120 meters.
   - Communication laptops to smartphone for audio communication can be done up to 200 meters, but for video communication only at a distance about 80 meters.
   - Communication between Smartphones for audio communication can be done up to a distance of 140 meters and for video only up to a distance of 80 meters.
5) Based on ITU-G.114 Standard on delay, both laptop to laptop and laptop to smartphone communications can be done well up to 200 meters.

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