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Portable and Reliable Telecommunication System on IP PBX in Wireless Mesh Network

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Abstract —The aim of the research is to design a portable wireless mesh networks based on IP PBX technology. The energy source of the system is supplied from the combination of solar cell and battery. The system employs Asterisk FreePBX software as a server configuration media while the clients, laptop and smartphone device, employ MicroSIP and Bria as a softphone. 13 concurrent calls perform 21.49% of CPU utilization, outperforming the previous research of around 70%. The results also describe that all the network performance parameters such us delay, jitter, and packet loss have met the ITU G.114 standards.

Index Terms—IP PBX, Raspberry Pi, VoIP, Wireless Mesh Network

I. Introduction

Current telecommunication infrastructure has not yet covered every area, especially in rural areas. Besides requiring high fund, the development of communication infrastructure also takes relatively long time. Therefore, it has been considered to build a telecommunication system that is affordable and can be built with in short time and should allow independent usage. Voice over Internet Protocol (VoIP) is one of the current communication technologies that rapidly grows. VoIP is an IP-based telecommunication technology. The technology can use a set of minicomputer like Raspberry pi as server and access point which functions as network provider, making this technology flexible and portable.

Raspberry Pi is a mini computer product for various purposes. Despite its tiny size, this device offers reliable functions. The device can operate in various operating systems including linux, windows, even android. Compared to other types of computer, raspberry pi is superior in term of size, making this device portable and it requires relatively lower wattage. Hence, Raspberry Pi can be used as VoIP server using battery as power input.

There have been several previous researches on VoIP-based communication. A research done by [1] - [5] show that Raspberry Pi can be utilized in VoIP real time communication for voice and image messages. In a research conducted by [6], a communication system was designed using two mini computers as IP PBX-based. This research has successfully increased the range that can be reached by the system up to 200 meters using WLAN mode Extended Service Set network topology. It was found in the research of [1] that the 70% of CPU utilization of server serve only 12 concurrent calls. All of those previous researches found the device to suffer from limited coverage area and limited capability of the server in handling concurrent calls.

In this research, the wireless mesh network topology was employed to improve the network performance. The application of mesh topology in wireless network has been known to improve the quality of concurrent calls and allow better network recovery mechanism when the calls are disrupted in other networks. This research also employed solar energy as power supply, making this device usable in remote areas which have not yet been equipped with electricity and areas which facing electricity problems such as the condition after disasters.

II. Literature Review

A. Raspberry Pi

Raspberry Pi 3 has a 1.2 GHz Quad-Core ARMv8 processor (64Bit) system on chip (SOC), 802.11n wireless LAN, and Bluetooth low energy (BLE) 4.1. It can be used in some electronic projects and many things that desktop PCs usually do. The Raspberry Pi board also contains a graphics chip dual core video core IV, 1GB LPDDR2 RAM, on chip antenna for wireless LAN and Bluetooth, and various interfaces for external devices. Its operating system boots on Micro SD card, can run in windows or Linux. It is powerful, small (85 x 56 x 17mm), low power consume and cheap [7]. Physically, Raspberry Pi 3 can be seen in Fig. 1.

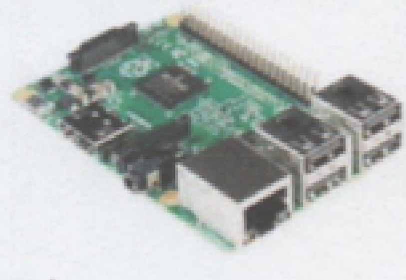

Fig. 1. Raspberry pi

A. Voice Over Internet Protocol (VoIP)

VoIP is a voice call technology that changes analog data into digital data [1]. Voice data is converted into digital code and then sent as data packages through

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internet media. VoIP is also known as internet telephony, IP telephony or digital telephony. VoIP provides large bandwidth capacity that supports voice and video conference [8]. The basic concept of VoIP is described in Fig. 2

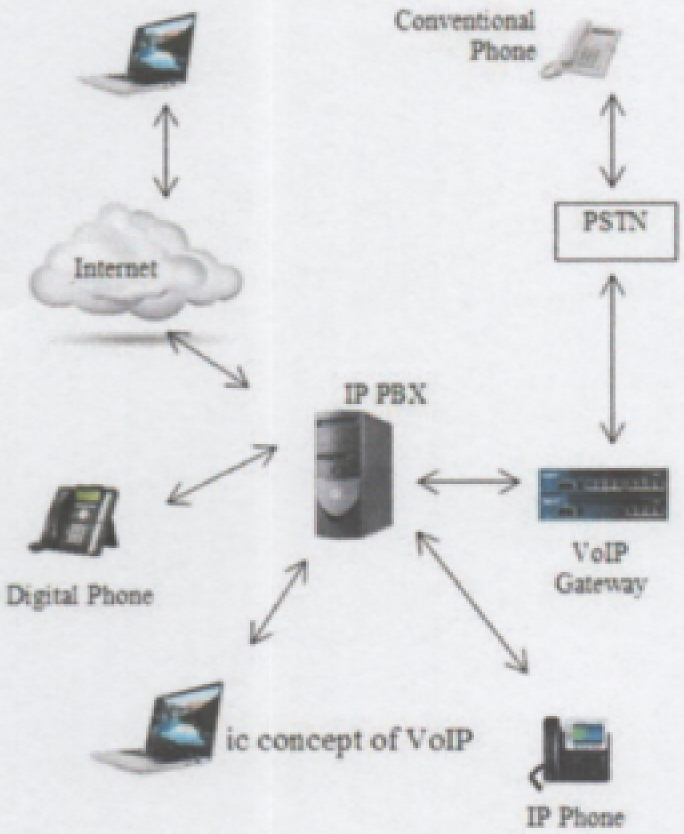

Fig. 2. Basic concept of VoIP

VoIP implements a signalling protocol to support the communication between users. The signalling protocols work for setting up and tearing down a call. The signalling protocols are:

- H.323
- Session Initiation Protocol (SIP)
- Media Gateway Control Protocol (MGCP)
- Inter-Asterisk eXchange (IAX)
- Session Description Protocol (SDP)
- Real-time Transport Protocol (RTP)
- Jingle XMPP VoIP extensions

B. *Private Automatic Branch eXchange (PABX)*

PABX is a central call switching with some extensions that can be used by every user. Each user is given an extension number which allows the outcoming calls to be directed to the extension number. Each user can also communicate by dialing the extension numbers. PABX typically provide some features to make information exchange more convenient.

C. *Internet Protocol Private Branch eXchange*

Internet Protocol Private Branch Exchange (IP PBX) is call switching device based on Internet Protocol (IP) technology using analog telephone as well as IP Phone extension. This device provides various functions including the ones related to connection, control, phone disconnection, communication protocol translation, communication media translation, even control upon IP-based telephone devices such as VoIP Gateway, Access Gateway and Trunk Gateway [9], [10].

D. *Asterisk FreePBX*

Asterisk is an open source application used in telecommunication system [8]. Asterisk can be configured in network that runs on IP, PBX hybrid or phone switching systems. Asterisk can also be used in router management as it activates several features and connects a caller to IP from the outside to analog and digital connections. Asterisk also runs on various operating system including Linux, Mac OS X, OpenBSD, FreeBSD and Sun Solaris that provides various features needed by PBX.

E. *Wireless Mesh Network*

Wireless Mesh Network (WMN) is a wireless communication network topology that is formed by nodes, in which each node has two or more communication lines. Each node not only serve as a host but it also functions as router to forward the information packets to other nodes. The main characteristic of wireless mesh network is its capability in configuring and organizing itself, or in other words, it is able to maintain its connectivity when a node is disrupted [11], [12]. WMN promises high flexibility, reliability and performance over conventional WLANs. The implementation of WMN technology can improve network scalability [13].

F. *Network Performance Parameters*

Some disruptions might occur to the data packet before the data reach the destination. Therefore, performance of network such us throughput, delay, jitter, and packet loss should be measured to know the quality of the networks [14]-[17]. The recommendation standard of the network performances set by International Telecommunication Union - Telecommunication (ITU-T) G.114 are presented in Table I.

TABLE I: ITU-G.114 RECOMMENDATION STANDARD

Delay (ms)	*Jitter (ms)*	*Packet Loss (%)*	*Quality*
0 – 150	*0 - 20*	*0 – 1*	*Good*
150 – 400	*20 – 50*	*1 – 5*	*Fair*
> 400	*> 50*	*> 5*	*Poor*

Delay

Delay refers to the accumulation of delay time in the network from the time the data packet is sent up to the time the packet data is received. Delay affects the quality of services as it takes longer for the data packet to be received. ITU-T G.114 recommends that delay should not exceed 150 ms in various applications.

Jitter

Jitter is the variation in the time interval of arrival for a data packet at the receiver. The variation of the arrival time might be caused by congestion in networking, improper queuing, weak network capacity, size of the data, and the accumulation of the time delay of each packet. The delay and jitter are important problems as it effect in degradation of data packet quality.

Packet loss

Packet loss is defined as the total of lost packet or the ones that cannot reach the receiver. The loss of the packet is caused by several factors including overload traffic, collusion, error in the devices, and overflow in the buffer.

Throughput

Throughput is an actual transfer rate between source and destination, measured in bits per second (bps) or bytes per seconds (Bps). The throughput represents the average speed of data received by destination node at certain networks time during sending and receiving data.

III. System Design

A. General Overview of the System

The basic concept of this system design is providing IP PBX-based wireless mesh communication network using minicomputer Raspberry Pi as the server. This system facilitates both voice call and video call. Fig. 3 shows the flowchart of the system.

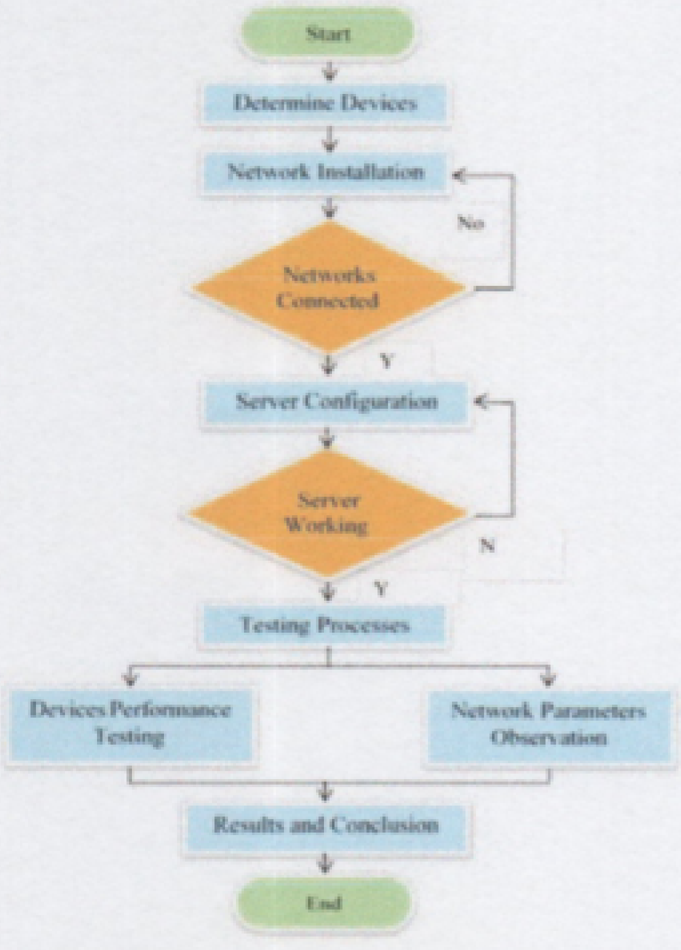

Fig. 3. Basic concep of the system design

B. Network Topology

This system applies wireless mesh network topology in which each server has different IP address. Fig. 4 presents the system topology developed in this research.

IV. Results

A. The Test on Mesh Network Topology

The test was done by communicating between the servers at the same time. The topology of the testing is presented in Fig. 4. The testing on this mesh topology was in the form of ping communication among clients using putty application. The results of the test are presented in Table II.

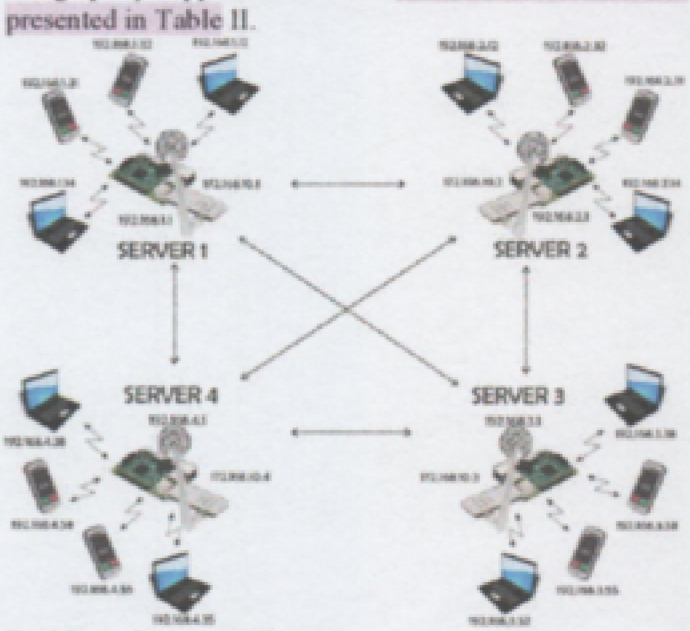

Fig. 4 Network system topology

TABLE II: MESH NETWORK TOPOLOGY

No.	Source Server	Destination Server	Remarks
1	Server 1 (192.168. 1.1)	Server 3 (192.168.3.1)	Success
2	Server 1 (192.168.1.1)	Server 4 (192.168.4.1)	Success
3	Server 2 (192.168. 2.1)	Server 3 (192.168.3.1)	Success
4	Server 2 (192.168.2.1)	Server 4 (192.168.4.1)	Success
5	Server 3 (192.168.3.1)	Server 1 (192.168.1.1)	Success
6	Server 3 (192.168. 3.1)	Server 2 (192.168.2.1)	Success
7	Server 4 (192.168.4.1)	Server 1 (192.168.1.1)	Success

B. The Test on CPU Performance in Server.

The test was conducted by making several calls at the same time while the performance of the CPU in each server was being observed. There were 4 servers and some clients involved in this test, which results can be seen in Table III.

C. The Test on Quality of Call in Server

There are four scenarios applied to measure the call quality of the networks. They are:

- Communication between smartphone (SP) and smartphone (SP).
- Communication between laptop (LT) and smartphone (SP).
- Communication between smartphone (SP) and laptop (LT).

- Communication between laptop (LT) and laptop (LT).

Voice calls employ the PCMA or G.117 codecs while video calls use the H.264 codec with VGA image resolution (640 x 480 pixels). Measurement of call performance parameters on laptops was carried out using Wireshark application. Tests were carried out within Line of Sight (LOS) condition for a duration of 120 s for each [10]. The call performance parameters observed included delay, jitter, throughput, and packet loss. The measurement results for the scenarios test can be seen in Table IV – Table XI.

D. The Test on Solar Panel

The test on solar panel was conducted in outdoor area during shiny day from 07.00 AM to 17.00 PM. The solar panel used is the SunPower Model BW-L1 with power 20WP. In the test, the output of both voltage and current of the device were observed, which the observation results are presented in Table XII.

V. Discussions

A. Analysis of CPU Performance

Fig. 5 shows that the number of concurrent calls affected the CPU performance in the servers. The more calls in each server trigger the higher CPU server usage. It can be seen in Table III that there is an increase in CPU usage when 13 concurrent calls with an average percentage of 21.49%.

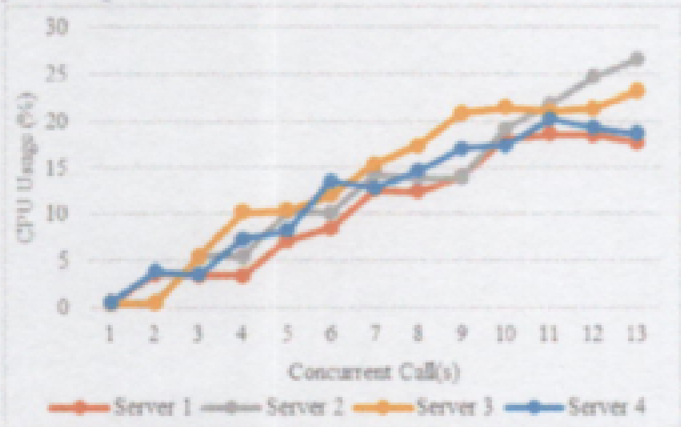

Fig. 5. CPU performance

TABLE III: CPU PERFORMANCE

Concurrent Calls	CPU Usage (%)			
	Server 1	Server 2	Server 3	Server 4
1	0.38	0.46	0.46	0.54
2	3.62	0.54	0.46	3.84
3	3.40	5.40	5.38	3.48
4	3.42	5.40	10.22	7.24
5	7.06	10.22	10.40	8.20
6	8.44	10.08	12.04	13.48
7	12.58	14.22	15.28	12.82
8	12.38	13.90	17.34	14.58
9	14.00	13.90	20.84	17.04
10	18.04	19.14	21.42	17.36
11	18.60	21.84	21.02	20.20
12	18.48	24.68	21.34	19.28
13	17.66	26.54	23.16	18.6

TABLE IV: PERFORMANCE PARAMETERS OF LT-LT FOR VOICE COMMUNICATION

Concurrent Calls	Delay (ms)	Jitter (ms)	Packet loss (%)	Throughput (Kbps)
	Voice	Voice	Voice	Voice
1	0.02	5.70	0.10	8.55
2	0.02	7.75	1.00	13.89
3	0.02	10.24	1.80	15.08
Average	0.02	7.89	0.97	12.51

TABLE V: PERFORMANCE PARAMETERS OF LT-LT FOR VIDEO COMMUNICATION

Concurrent Calls	Delay (ms)	Jitter (ms)	Packet loss (%)	Throughput (Kbps)
	Video	Video	Video	Video
1	0.01	14.82	1.50	8.03
2	0.02	17.61	2.00	46.91
3	0.01	22.64	4.90	95.73
Average	0.01	18.36	2.80	50.22

TABLE VI: PERFORMANCE PARAMETERS OF LT-SP FOR VOICE COMMUNICATION

Concurrent Calls	Delay (ms)	Jitter (ms)	Packet loss (%)	Throughput (Kbps)
	Voice	Voice	Voice	Video
1	3.00	9.10	0.10	40.50
2	4.00	10.90	1.00	48.30
3	3.50	11.10	0.20	58.40
Average	3.50	10.37	0.43	49.07

TABLE VII: PERFORMANCE PARAMETERS OF LT-SP FOR VIDEO COMMUNICATION

Concurrent Calls	Delay (ms)	Jitter (ms)	Packet loss (%)	Throughput (Kbps)
	Video	Video	Video	Video
1	8.00	13.71	0.00	40.50
2	8.50	15.30	0.00	48.30
3	16.8	22.00	3.30	58.40
Average	11.1	17.00	1.10	49.07

TABLE VIII: PERFORMANCE PARAMETERS OF SP-LT FOR VOICE COMMUNICATION

Concurrent Calls	Delay (ms)	Jitter (ms)	Packet loss (%)	Throughput (Kbps)
	Voice	Voice	Voice	Voice
1	0.02	2.46	3.00	11.55
2	0.02	3.79	3.12	17.10
3	0.02	11.10	3.40	26.47
Average	0.02	5.78	3.17	18.37

TABLE IX: PERFORMANCE PARAMETERS OF LT-SP FOR VIDEO COMMUNICATION

Concurrent Calls	Delay (ms)	Jitter (ms)	Packet loss (%)	Throughput (Kbps)
	Video	Video	Video	Video
1	0.02	15.36	0.00	49.55
2	0.02	8.73	0.00	59.67
3	0.02	16.44	3.19	95.84
Average	0.02	13.51	1.06	68.35

TABLE X: PERFORMANCE OF SP-SP FOR VOICE COMMUNICATION

Concurrent Calls	Delay (ms)	Jitter (ms)	Packet loss (%)	Throughput (Kbps)
	Voice	Voice	Voice	Voice
1	4.00	12.10	0.10	42.90
2	3.50	13.20	1.30	34.10
3	3.00	17.00	4.60	82.30
Average	3.50	14.10	2.00	53.10

TABLE XI: PERFORMANCE OF FOR SP-SP VIDEO COMMUNICATION

Concurrent Calls	Delay (ms)	Jitter (ms)	Packet loss (%)	Throughput (Kbps)
	Video	Video	Video	Video
1	75.8	11.50	0.00	76.70
2	70.9	14.40	0.00	79.90
3	76.2	33.10	1.90	79.90
Average	74.3	19.67	0.63	78.83

TABLE XII: SOLAR PANEL TESTING

No.	Time Observations	Voltage (Volt)	Current (A)
1	07:00 AM	4.51	0.08
2	07:30 AM	4.57	0.09
3	08:00 AM	5.19	0.55
4	08:30 AM	5.7300	0.73
5	09:00 AM	5.20	1.30
6	09:30 AM	5.30	1.50
7	10:00 AM	6.00	1.69
8	10:30 AM	6.73	1.80
9	11:00 AM	6.89	1.80
10	11:30 AM	11.0	1.80
11	12:00 PM	12.0	1.80
12	12:30 PM	13.0	1.70
13	13:00 PM	11.0	1.00
14	13:30 PM	7.00	1.00
15	14:00 PM	5.61	1.08
16	14:30 PM	4.61	0.17
17	15:00 PM	4.85	0.20
18	15:30 PM	4.85	0.08
19	16:00 PM	4.18	0.20
20	16:30 PM	4.56	0.08
21	17:00 PM	4.57	0.08

B. *Analysis on Call Performance*

Delay

Based on Fig. 6, voice communication with laptop and laptop scenarios show the least delay value for both voice and video calls of 0.02ms and 0.01ms respectively. The highest delay occurred in Laptop to Smartphone communication for video calls with a value of around 11.10 ms. The delay values still meet the standards required by ITU-G114.

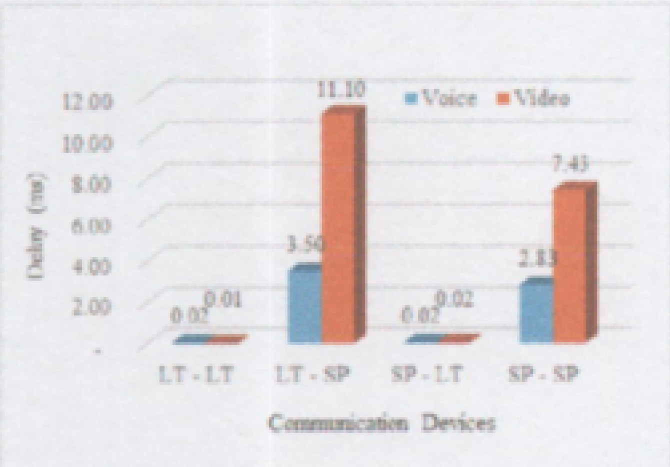

Fig. 6. Average delay for voice and video communications

Jitter

Based on Fig. 7, all types of scenarios show that voice call has a jitter value smaller than the one in video call. The biggest jitter value was found in the communication between Smartphone and Smartphone for video communication at 19.67 ms. The jitter value is quite big compared to the jitter value for voice communication of 4.43ms. However, those jitter values are smaller than 20 ms which considered good according to ITU-G114 standards.

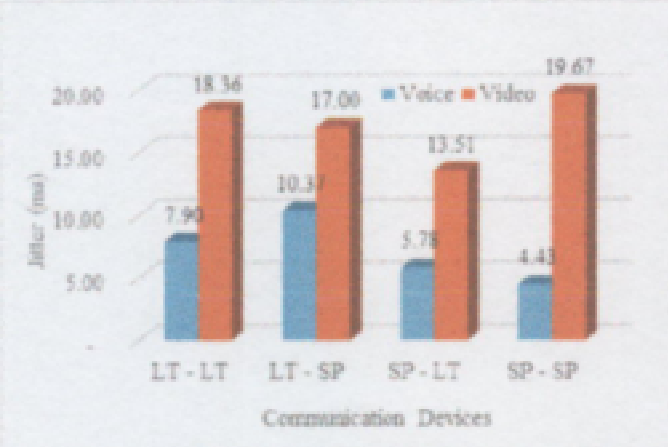

Fig. 7. Average Jitter for voice and video communications

Packet loss

Fig. 8 presents the average value of packet loss in voice communication, ranging from 0.27% - 0.97%. The values meet the good standard by ITU-G114. Video communication has a higher value of packet loss from 0.63% for communication between smartphones to 2.80% for communication between laptops.

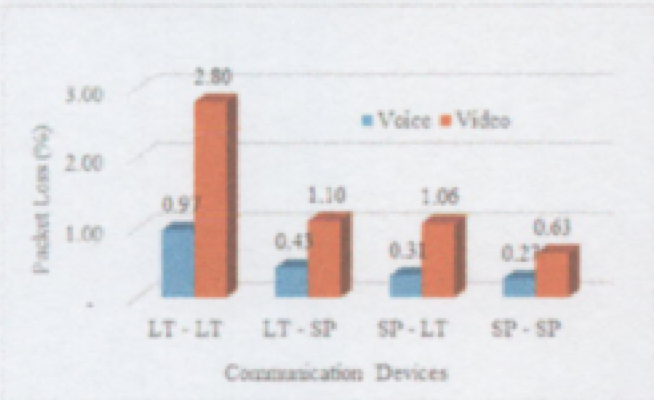

Fig. 8. Average packet loss for voice and video communications

Throughput

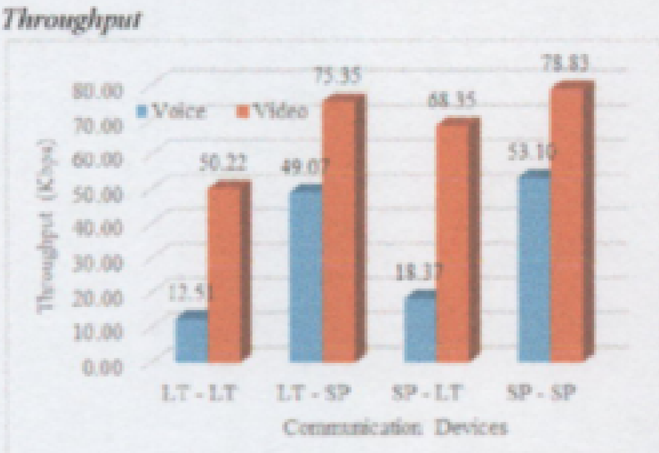

Fig. 9. Average throughput for voice and video communication

Fig. 9 illustrates the average value of throughput for voice and video communication. The average throughput varies between 50.22 Kbps and 78.83 Kbps for video communication. The voice communication between laptop and laptop has a smallest average value of throughput at 12.51 Kbps.

C. *Analysis of Solar Panel*

Solar panel testing was carried out between 07.00 AM and 05.00 PM. During the time, the power bank battery of solar panels was recharged. The output of the solar cell testing can be seen in Table XII. After 5 PM, the power for the server devices were supplied by the battery. Based on the test results on the servers, the system ran normally supplied by solar panel and battery. Therefore, the system can be used in areas without electricity or areas that experience electricity problems such as natural disaster areas.

V. Conclusions

A. IP PBX-based telecommunications system developed in this research can be used for both voice and video calls using wireless mesh network topology.

B. 13 concurrent calls took up to 21.49% of CPU resource.

C. The highest values of delay, jitter, packet loss, and throughput occurred in voice calls of 3.50 ms, 10.37 ms, 0.97%, and 53,10 Kbps respectively. While in video calls, the largest average value of delay, jitter, packet loss, and throughput is 11.10 ms, 19.67 ms, 2.80%, and 78.83 Kbps respectively. All communication performance parameters meet the standard communication requirements set by ITU-T G.114.

Acknowledgment

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ORIGINALITY REPORT

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SIMILARITY INDEX	INTERNET SOURCES	PUBLICATIONS	STUDENT PAPERS

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