IMPLEMENTATION OF VVOIP SERVER BASED ON RASPBERRY PI USING WEB RTC AS API

Muhammad Aufa Khodhi Ilyasa, Jakarta State Polytechnic

***Abstract*—In this study Video conferencing applications that are currently developing such as Zoom and Google Meet have an important role in human productive activities, such as working, studying, holding meetings and others. Video conferencing technology is a technology that can bring together two or more people with sound and visual appearance that is broadcast live. Rspberry Pi is a portable server that capable to handle voice and image processing quite good with the values of all QoS tested.**

Concluded that the system that has been built has been successful with a success percentage of VoIP calls toVoIP is 100%., the percentage of success. 100%.. In terms of QOS, the system has shown good quality, meeting the communication standards set by ITU-T, both in terms of jitter, packet loss, delay and throughput. The maximum range of VoIP client to server in order to make VoIP to VoIP calls properly is 100 meters in Line Of Sight (LOS) conditions. At a distance of 150 meters can still communicate but not stable. Meanwhile, the VoIP client to server range is 25 meters and in NLOS conditions, VOIP communication/calls can still be made. In NLOS conditions, connection loss occurs at a distance of 30 meters.

***Index Terms*—Call conference, Video call, Raspberry Pi, Mikrotik, Asterisk, WebRTC**

I. Introduction

Recently, During the current covid pandemic, long-distance communication is increasing. Video conferencing applications that are currently developing such as Zoom and Google Meet have an important role in human productive activities, such as working, studying, holding meetings and others (Yansen Theopilus, 2021). Video conferencing technology is a technology that can bring together two or more people with sound and visual appearance that is broadcast live.

In running a conference call system, a capable server is needed to process voice and video data so that it can run smoothly. In this study, a Raspberry Pi 4B will be used as a server and a Mikrotik RB 941 router as a network controller. Which will later be developed into a video conference system that is reliable and has good performance quality (Agnieszka Chodorek, 2021)

The Raspberry Pi 4B was chosen because it has 3 times better performance capabilities than its predecessor (3B, 3B+) (Halfacree, 2019). As well as adding limitations to research (Fatma Salih, 2018) and (S. Neha Vimala, 2017) which ran out of RAM and Processor resources. Also apply it to LAN and WLAN networks with up to 5 users

In supporting video conferencing activities, it is necessary to display a good application that is easy to use. WEB RTC offers good transmission of video and audio streams, and ensures good data flow protection and has an almost real-time character that allows contextual communication (Agnieszka Chodorek, 2021). The use of WEB RTC is also based on research (Ida Bagus Ary Indra Iswara, 2021) which is considered quite good in displaying video conference servers.

This study will focus on the implementation and analysis of QoS performance on a VVoIP server based on Raspberry Pi 4B and using WEB RTC as an API on CV Varia Ikhtiara Prima.

Tests in this study will focus on the QoS performance of the server in the form of: Delay, Throughput, Jitter and Packet Loss. Testing is done using wireshark tools.

From this background description, it will be discussed about the implementation of VVoIP Server, QoS

This paragraph of the first footnote will contain the date on which you submitted your paper for review, which is populated by IEEE. It is IEEE style to display support information, including sponsor and financial support acknowledgment, here and not in an acknowledgment section at the end of the article. For example, "This work was supported in part by the U.S. Department of Commerce under Grant BS123456." The name of the corresponding author appears after the financial information, e.g. *(Corresponding author: M. Smith).* Here you may also indicate if authors contributed equally or if there are co-first authors.

The next few paragraphs should contain the authors' current affiliations, including current address and e-mail. For example, First A. Author is with the National Institute of Standards and Technology, Boulder, CO 80305 USA (e-mail: author@ boulder.nist.gov).

Second B. Author, Jr., was with Rice University, Houston, TX 77005 USA. He is now with the Department of Physics, Colorado State University, Fort Collins, CO 80523 USA (e-mail: author@lamar.colostate.edu).

Third C. Author is with the Electrical Engineering Department, University of Colorado, Boulder, CO 80309 USA, on leave from the National Research Institute for Metals, Tsukuba 305-0047, Japan (e-mail: author@nrim.go.jp).

Mentions of supplemental materials and animal/human rights statements can be included here.

Color versions of one or more of the figures in this article are available online at http://ieeexplore.ieee.org

PERFORMANCE AND THE RESULTS OF THE ANALYSIS OF THE IMPLEMENTATION AND QOS PERFORMANCE THAT WILL BE STUDIED.

II. LITERARY REVIEW

Session Initiation Protocol (SIP) is a signaling protocol for video conferencing, VoIP, multiplayer gaming and chat applications. It lays down a set of rules for how two systems communicate with each other by initiating a session. It is a method intended for communication with different location points [3].The Raspberry-Pi is a series of credit card-sized single-board computers developed in the United Kingdom by the Raspberry Pi Foundation. The Raspberry Pi 4B has the following specifications:
1. Processor : Broadcom BCM2711, Quad core Cortex-A72 (ARM v8) 64 bit SoC @ 1.5GHz
2. RAM: 4GB
3. Connectivity : 2.4 GHz and 5.0 GHz IEEE 802.11ac wireless, Bluetooth 5.0, BLE Gigabit Ethernet
4. Ports: 2 USB 3.0 ports; 2 USB 2.0 ports.
Raspberry Pi standard 40 pin GPIO header (fully backward compatible with previous boards)
2× micro-HDMI ports (up to 4kp60 supported)
2-lane MIPI DSI display port
2-lane MIPI CSI camera port
4-pole stereo audio and composite video port

5. Power : 5V DC via USB-C connector (minimum 3A*)
5V DC via GPIO header (minimum 3A*) (see Fig. 1). WebRTC is a framework that brings together several established protocols as shown in Figure 2.1. Uses peer-to-peer connections for arbitrary video, audio and data transmission. The aim of this framework is to provide real-time capabilities for application developers through a simple and standard interface with the prevalence of web software and the ability to create rich and diverse applications in browsers [4](see Fig.2). Asterisk is an IP PBX software to create a telephone communication service system over the internet or commonly called VoIP (Voice over Internet Protocol). Asterisk is an open source PBX software that runs on a Linux-based operating system. Acts as a VoIP server software which is distributed through the GPL (General Public License). Asterisk is also called IP PBX, because it has functions and capabilities like a PBX but is IP-based. Asterisk is usually used to build a communication service system and makes it easy to develop your own communication and telephone services with extensive customization provided to the user with a symbol that represents a widcard in many computer languages [2]. Wireshark is a network analysis tool, also known as protocol analysis or packet sniffer. Wireshark can be used for network troubleshooting, analysis and development for educational purposes. Wireshark works by capturing data running through the NIC (Network Interface Card). The data obtained is the data used to measure the performance of the video conferencing system to be built. [1]. The test parameters in this study were recorded using Wireshark software against several Quality of Services (QoS) parameters.
Quality of Service is the ability of a service to guarantee the performance of a system that will be recorded and analyzed including throughput, packet loss, delay, and jitter [8]. There are Quality of Services (QoS) standards, one of which is TIPHON (Telecomunications and Internet Protocol Harmonization Over Network) TR.101329.V2.1.1.1999-06 issued by ETSI (European Telecommunications Standards Institude). (see table 3)

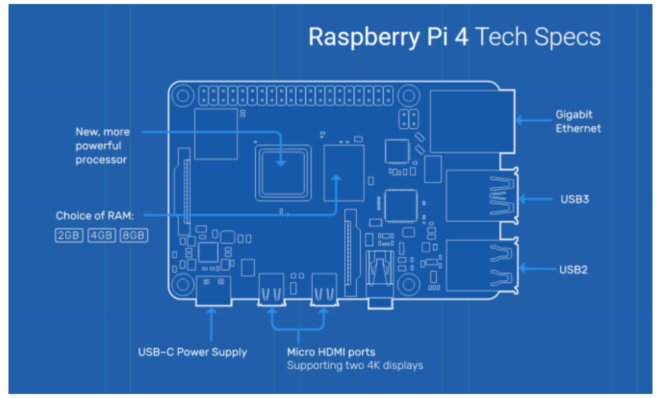

Fig. 1. Raspberry Pi 4B Hardware

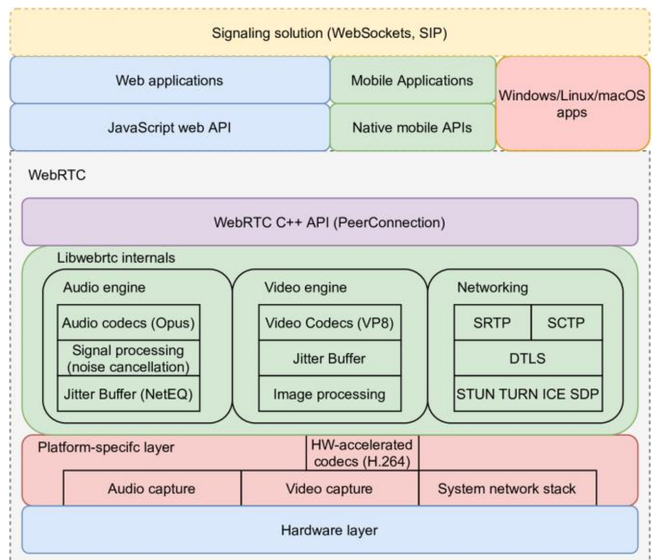

Fig. 2. WebRTC Structure

Index Value	Presentate	Category
3,8 – 4	95 – 100%	Very Good
3 – 3,79	75 – 94,75%	Good
2 – 2,99	50 – 74,75%	Medium
1 – 1,99	25 – 49.75%	Bad

Fig. 3. Quality Of Service Index

Mikrotik is a computer network device in the form of hardware and software that functions as a router, network filter and switching device [6]. The router in this study serves as network monitoring, and setting IP sharing on the device. The Mikrotik router used in this study is the hAP lite series (RB941-2nD) with the following specifications:.(see Fig.4)

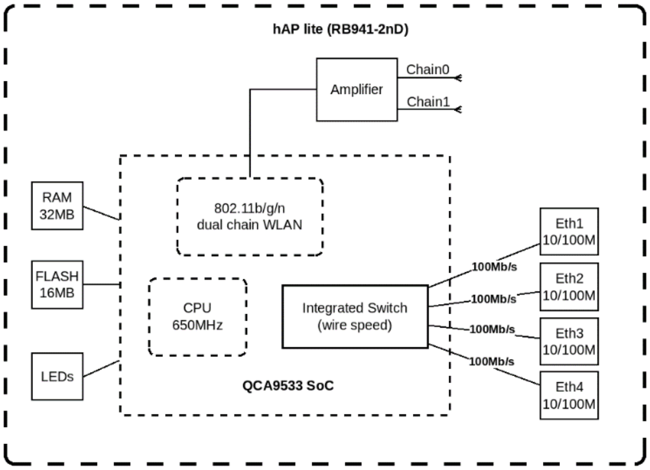

Figure. 4 Mikrotik Block Diagram

III. Methodology

A. Research design

This study uses a quantitative approach, according to [9] a research approach that uses a lot of numbers, starting from collecting data, interpreting the data obtained, and presenting the results. So this approach is suitable to be applied in this research. The type of research carried out is an observational case study because it conducts an assessment and analysis of certain objects so that research is carried out by going directly to the field to obtain existing data.

Data collection and analysis techniques are in the form of system QoS test results under certain conditions which will later be compared with related research, so that objective conclusions can be drawn..

B. Stages of Research

The stages of research carried out using the Network Development Life Cycle (NDLC) methodology.
Metodologi NDLC memiliki 6 tahapan dalam siklusnya :

1. Analysis

   At this stage, needs analysis, user desire analysis and network topology analysis will be carried out. With several methods such as: interviews, field surveys and documentation for later design materials

2. Design

   At this stage, a blueprint for the network topology and system design will be made based on the analysis data that has been done previously and the estimated costs needed to conduct research.

3. Simulation Prototyping

   Perform simulations using third-party applications to see the possibility of errors and see the effectiveness of the system to be made as well as presentation material to related parties before it will be implemented.

4. Implementation

   At this stage, everything that has been designed and simulated will be applied based on real conditions.

5. Monitoring

   At this stage everything that has been implemented will be monitored. Some of the parameters that will be observed are:
   a. Hardware infrastructure reliability
   b. QoS of the running network
   c. Take a network management approach (Network Management) (Setiawan, 2015)
   d. Doing documentation

6. Management

   At this stage, management will be carried out on users such as policies that can be arranged so as to create maintained reliability

C. Object of Research

The focus of this research is the results of the QoS system testing on the implementation of the Raspberry Pi-Based VVOIP Server which is applied to CV Varia Ikhtiara Prima under certain conditions. Which will be compared with related research and also research on the QoS performance of existing WebRTC applications. (Network specifications, data communication, Responder)

IV. Discussions

A. Requirements Analysis

Requirements analysis is an activity that aims to collect and analyze a list of requirements on the VVOIP Server system. Several methods were used for the data collection process, namely conducting interviews with company owners and conducting direct field observations. So, based on the analysis, it was found the need to support the system to run well..

B. System Design

Based on Figure 5, the system implementation scheme is as follows:
1. Users 1 and 2 make network connections on Mikrotik
2. Mikrotik is connected with an internet connection
3. Mikrotik router has been connected to raspberry pi with nip: 192.168.30.88
4. Connected Raspberry 4B is configured with WebRTC, Asterisk 16, and TLS Certification
5. Users 1 and 2 can make phone calls, chat and conduct video conferences in real time by opening the web application at https://192.168.30.88:443

(See Fig.5)

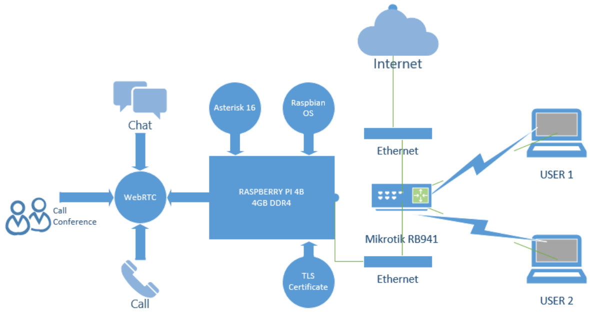

Fig. 5. System Diagram

Based on Figure 6, the raspberry pi is connected to the internet via port 1, so port 1 has a configuration as a DHCP Client which then distributes bandwidth with simple routing to 3 other ports along with a DHCP wireless interface. On port 3 where the Raspberry is located, it has been set to use Static IP so that the IP does not change.

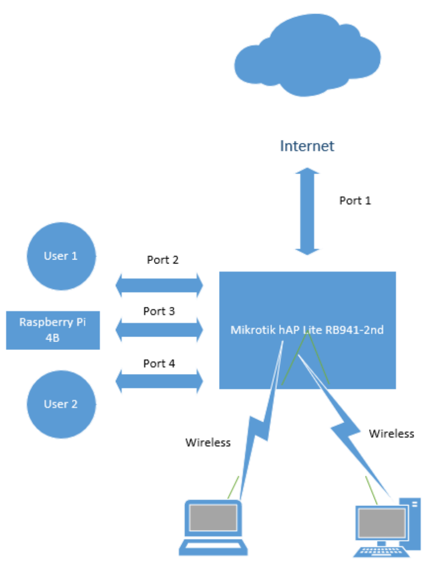

Fig. 6. Mikrotik Config

Below is the configuration for the sip.conf for the call protocol

```
  GNU nano 5.4                              sip.conf *
[general]

udpbindaddr=0.0.0.0:5060

context=from-outside
context=from-extension
useragent=Raspberry Pi
realm=192.168.30.88
srvlookup=no
mohsuggest=default
parkinglot=default
allowguest=no
alwaysauthreject=yes
videosupport=no
maxcallbitrate=5120
ignoreregexpire=no
allowsubscribe=yes
notifyhold=yes
notifyringing=yes
callcounter=yes
```

Fig. 7. Sip.conf Configuration

C. System Implementation

Based on Figure 5, the system implementation scheme is as follows:

1. Users 1 and 2 make network connections on Mikrotik
2. Mikrotik is connected with an internet connection
3. Mikrotik router has been connected to raspberry pi with nip: 192.168.30.88
4. Connected Raspberry 4B is configured with WebRTC, Asterisk 16, and TLS Certification
5. Users 1 and 2 can make phone calls, chat and conduct video conferences in real time by opening the web application at https://192.168.30.88:443

(See Fig.5)

D. Data Analyze

In the system testing section, QoS parameter monitoring will be carried out using Wireshark installed on the Raspberry Pi on the Ethernet interface. Tests are carried out to find out whether the Raspberry Pi has good QoS quality in running a VVOIP server using Asterisk

The test was carried out by making 3 voice calls and 3 video calls with a duration of 1 minute each. Then make observations on Wireshark.

There are 6 testing procedures to be carried out:

1. Scenario 1: Voice call for 1 minute
2. Scenario 2: Video call for 1 minute
3. Scenario 3: Call conference call 3 users for 1 minute

Below is the values of all test

Test	Packet Received (Bytes)	Observation Duration (Sec)	Throughput (bits/sec)	Average Througput
1	1,019,856	74.247	109,887.91	227,942.30
2	844,005	66.416	101,662.85	
3	3,813,544	68.002	448,639.04	
4	3,672,721	73.359	400,520.29	
5	630,135	63.81	79,001.41	

Fig. 8. Throughput Average

Test	Total *Packet*	Total Delay	Average Delay	Average Total Delay
1	7,169	74.25	10.36	8.35
2	7,028	65.38	9.30	
3	10,687	68.00	6.36	
4	11,072	73.36	6.63	
5	7,012	63.81	9.10	

Fig. 9. Delay Average

Test	*Packet* Dikirim	Packet Received	*Packetloss*	Average Packet loss (%)
1	7,169	7,163	0.08	23.20
2	7,048	6,652	5.52	
3	10,687	6,916	52.60	
4	11,072	7,453	50.48	
5	7,012	6,488	7.31	

Fig. 10. Packetloss

Test	Packet Received	Total *Jitter*	*Average Jitter*	Total rata rata jitter
1	7,169	0.00	0.00	0.00
2	7,029	0.00	0.00	
3	10,687	0.01	0.00	
4	11,072	0.00	0.00	
5	7,012	0.02	0.00	

Fig. 11. Jitter

Quality of Service

After testing the quality of service of the VPN connection using Wireshark, from 10 times, the average value of the QoS parameter is shown as (see Table. II).

Parameter	Value
Throughput	229.942 bits/sec
Delay	8.35 ms
Packet Loss	0%
Jitter	0.00 ms

V. Conclusion

Rspberry Pi is a portable server that capable to handle voice and image processing quite good with the values of all QoS tested. Concluded that the system that has been built has been successful with a success percentage of VoIP calls to VoIP is 100%., the percentage of success. 100%.. In terms of QOS, the system has shown good quality, meeting the communication standards set by ITU-T, both in terms of jitter, packet loss, delay and throughput. The maximum range of VoIP client to server in order to make VoIP to VoIP calls properly is 100 meters in Line Of Sight (LOS) conditions. At a distance of 150 meters can still communicate but not stable. Meanwhile, the VoIP client to server range is 25 meters and in NLOS conditions, VOIP communication/calls can still be made. In NLOS conditions, connection loss occurs at a distance of 30 meters.

References

[1] Bagus Ary Indra Iswara, I., Putu Pedro Kastika Yasa, I., n.d.

[2] Dwiyatno, S., Nugraheni, M., 2019

[3] Parmar, N., Ranga, V., 2019. International Journal of Innovative Technology and Exploring Engineering 8, 679–686.

[4] Tanskanen, S., Antti, S.P., Advisor, Y.-J., Severi Tikkala, M., 2021. Latency contributors in WebRTC-based remote control system..

[5] Teten Dian Hakim, Muryadi, 2019. Jurnal Ilmiah Elektronika 35–44.

[6] Ulum, F., 2018. DESAIN KEAMANAN JARINGAN PADA MIKROTIK ROUTER OS MENGGUNAKAN METODE PORT KNOCKING, Jurnal TEKNOINFO..

[7] Agnieszka Chodorek, R. R. C. S., 2021. UAV-Based and WebRTC-Based Open Universal Framework to Monitor Urban and Industrial Areas. SENSORS, pp. 1-35..

[8] Ardi Windiarto, K. W., 2019. The Design of Voice Over IP and GSM Gateway Using Raspberry Pi. TELKA, Volume 5, pp. 55-64..

[9] Arikunto, 2022. *Mengenal Perbedaan Penelitian Kualitatif dan Kuantitatif, Ini Penjelasan Lengkapnya.* [Online]
Available at: https://www.merdeka.com/jateng/mengenal-perbedaan-penelitian-kualitatif-dan-kuantitatif-ini-penjelasan-lengkapnya-kln.html
[Accessed 07 07 2022]..

[10] Fatma Salih, M. S. O., 2018. Raspberry Pi As A Video Server. *International Conference on Computer, Control, Electrical, and Electronics Engineering (ICCCEEE),* p. 4.

[11] S. Neha Vimala, M. V. C. K. K., 2017. Live Video Streaming from Remote Location Using Raspberry Pi. Indian Journal of Science and Technology, pp. 1-10..

[12] Setiawan, D., 2015. Network Management dan Strategi Bisnis Perusahaan. s.l.:s.n.

[13] yansen theopilus, m. p., 2021. Customer Preferences and Satisfaction Analysis of Video Conference Application for Productive Activities Using Kano Mode. Jurnal Rekayasa Sistem dan Industri, pp. 71-83.