SOLSR Protocol Performance Analysis For Voip Application In Mesh Topology

Aditya Wijayanto
Department of Software Engineering
Telkom Institute of Technology Purwokerto
Purwokerto, Indonesia
aditya.wijayanto@ittelkom-pwt.ac.id

Rifki Adhitama
Department of Software Engineering
Telkom Institute of Technology Purwokerto
Purwokerto, Indonesia
rifki@ittelkom-pwt.ac.id

Auliya Burhanuddin
Department of Informatics
Telkom Institute of Technology Purwokerto
Purwokerto, Indonesia
auliya@ittelkom-pwt.ac.id

Abstract—**Indonesia has a high level of potential for natural disasters that can cause damage to infrastructure; on the other hand, communication is critical when a natural disaster occurs, which causes communication to be hampered. The solution to dealing with this problem is building Ad-hoc Network (Manet) technology for communication media. By building alternative VoIP-based communication on Raspberry Pi. The protocol used in VOIP (Voice Over Internet Protocol) based communication using Raspberry Pi is the SOLSR (Secure Optimized Link State Routing) protocol. The VoIP work system is voice passing through the internet network in the VoIP system workflow; two components are connected, the VoIP client and VoIP server, VoIP server as the center. On the VoIP, the server uses Raspberry Pi, and the client also uses Raspberry Pi and is registered via SIP to the server to communicate to other clients and test its performance utilizing QOS testing using SOLSR. The test scenario is done by measuring throughput, delay, jitter, packet loss. The measurement results will produce the performance of a VoIP-based communication system using the SOLSR protocol and determine the feasibility of the SOLSR protocol to be applied to a VoIP-based communication system for disaster management.**

Keywords—SOLSR, VoIP, Raspberry Pi, Infrastructure, QoS

I. INTRODUCTION

Indonesia is located on the equator with various morphologies from landmasses to high mountains and seen from its geographic location. Indonesia located between Australia, Asia, the Indian Ocean, and the Pacific Ocean. Thus, Indonesia is also known as the Pacific region of fire or the Pacific circle of fire, which is an area that often occurs disasters such as earthquakes and volcanic eruptions that surround the Pacific Ocean curvature. The area is shaped like a horseshoe and covers 40,000 km [1].

Based on the Meteorology, Climatology, and Geophysics Agency (BMKG), in 2020 only, Indonesia experienced 699 landslides, 526 floods, 444 tornado events, 18 tidal waves and seven volcanic eruptions. In addition, seismic data states that in 2020 there have been ten earthquakes. Meanwhile, BMKG explained that the drought in 2020 impacted the agricultural sector, water resources, forestry, and the environment [2].

Indonesia has a high level of potential for disasters that cause damage to the communication infrastructure; communication is essential in everyday life. In times of disaster, communication is crucial. When natural disasters occur, people still use satellite phones as alternative communication. Satellite phones work significantly differently from ordinary cell phones; satellite phones deliver call signals to the satellite, then reflected into the ground. The cost of using this satellite phone is relatively expensive. The device price starts from IDR 8 million to IDR 25 million with a per-minute call rate of up to 1 US dollar or around 14 thousand rupiah.

There is a solution to overcome this condition and the effects of natural disasters, which often damaged infrastructure and rise into obstruction of communication by utilizing current technology using Mobile Ad-Hoc (Manet) by utilizing Raspberry Pi as a tool using SOLSR protocol. Manet is a network technology formed from a collection of ad-hoc nodes connected using a wireless connection transmission directly. However, if one node is outside the transmission range or down, it requires another node to forward messages from that node.

VoIP (Voice Over Internet Protocol) is a telecommunication network technology that can pass communication services over the internet protocol network to allow users to communicate or from server to client to exchange information in the form of a voice in the IP network. Using a VoIP network is efficient in bandwidth and management costs by utilizing Raspberry Pi as a server as an alternative solution for disaster-prone areas that allow long-distance conversations via internet media. By utilizing the working method of VoIP, voice data is converted into digital code and received through the network that sends data packets [3].

Raspberry Pi is a single-board computer made by the Raspberry Pi foundation. The first version of Raspberry has only 1 USB hub, an HDMI port, and a micro sd slot for storing the operating system. Then Raspberry Pi began to change to the size of a credit card and collaborate with Linux, which made the operating system and allow it to run on 700mhz, arm1, 176jzf-s processors. On the Raspberry Pi 4, there is a significant increase in the Bluetooth and Ethernet sectors. The Raspberry Pi 4 already uses the latest Bluetooth technology - Bluetooth 5.0. Bluetooth 5.0 has increased data transfer rates, reaching 2 Mbps (Bluetooth 4.2 - 1 Mbps), a further data transmission distance of up to 200 meters LoS (Bluetooth 4.2 - 50 meters LoS), and optimized for IoT applications. Then in the Ethernet section, both the Raspberry Pi 3B + and the Raspberry Pi 4 use Gigabyte Ethernet [3].

This communication system is designed to help the community as an alternative tool when the communication infrastructure is damaged so everybody can still be connected where there is minimum electricity supply and infrastructure by utilizing the voice model and using the nodes on each manet to communicate.

II. Methodology

In this study, we implement a VoIP communication network using Raspberry Pi in SOLSR (Secure Optimized Link State Routing) and analyze its performance. The communication architecture of proposed research described in Figure 1.

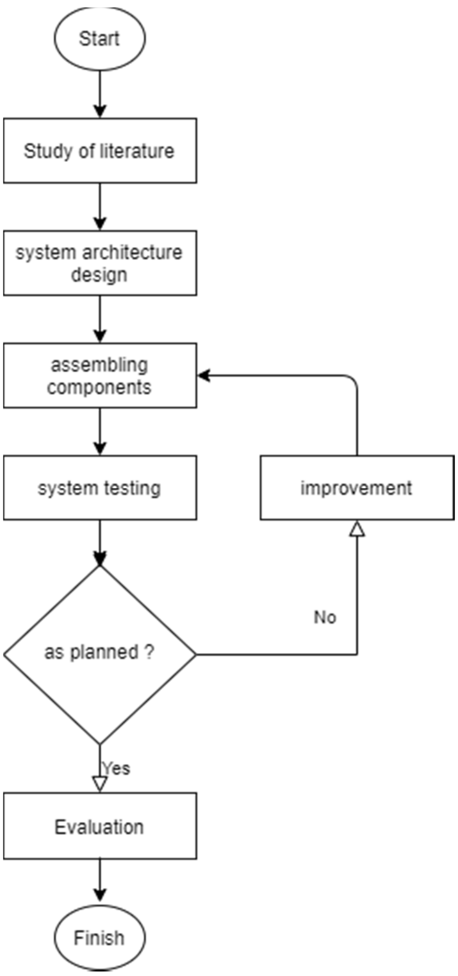

Figure 1. Research Methodology

A. General Communication System Architecture

The research begins by determining the number of nodes used to build the network and the routing protocol that will be used. In this system, Raspberry Pi work as a node and use the SOLSR protocol.

Figure 2. The System Architecture

Figure 2. Described Node ID 001 check for whether any devices have been formed. When Node ID 001 did not find any network that was formed, then Node ID 001 will become a server node, when another node comes, the server starts to form a network. Figure 2 also shows a network system that automatically changes, when node ID 004 enters the network it will form a new network and when Node ID 002 terminated from the network, the server repeats the configuration process from the beginning with a new number of nodes.

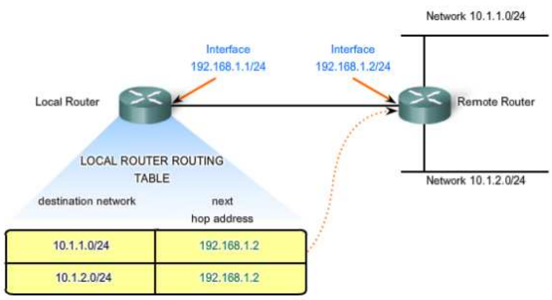

Figure 3. Visualisasi Informasi Pada Router Untuk Melakukan Routing [4].

Routing is a mechanism for determining the link process from the sending node to the receiving node that works at layer 3 Open Systems Interconnection (OSI) [5]. Routing is the process of moving data from one network to another by forwarding data packets via the gateway. Routing determines where data packets will be sent to reach the desired destination. A router learns the routing information from which its source and destination are then placed in the routing table. The router will rely on the table forward the packet to the destination address. Figure 3 below visualizes the information needed by a router to send data packets to different networks.

B. VoIP Communication System Scenario

Two main components are interconnected: the VoIP server and the VoIP client, the VoIP server as the center, and the VoIP client as the receiver. This uses a registered Raspberry Pi to SIP server to make calls via other VoIP clients [6]

The design of a VoIP communication system begins with determining the steps to build a manet network and routing protocols using VoIP to be used. The VoIP communication flow on the Raspberry Pi will explain using the SOLSR protocol in testing this system.

First steps: Run the asterisk SOLSR routing configuration. If connected, the next step check whether the SIP extension match the data in RasPBX [7].

C. Communication System Configuration

This section will explain several things that need to be configured both in the Ad-Hoc network configuration on each node and the SOLSR configuration.

- Ad-Hoc Network

  After the installation of the Raspbian buster operating system is successful, then the Ad-Hoc network configuration is carried out on each node to be built. Nodes in the network built using IPv4 and configured in the / etc / network / interfaces file. IP configuration is required so that each node can communicate with the other.

- OSLR

  The routing protocol used in this research is the OLSRv1 routing protocol (OLSR Daemon) by adding the security plugin. After installing OLSRv1 (OLSR Daemon), configure it in the olsrd.conf file to activate the SOLSR routing protocol.

- Supporting application

  The SOLSR routing protocol requires a supporting application to facilitate SOLSR installation and run SOLSR to make it more compatible. the supporting application in this research are kernel packages, libncurses5-dev, fakeroot, Wget, bzip2, g++, libssl-dev, Doxygen, Bison, Flex, libc6, libxml2-dev, voip asterisk, iperf, twinkle.

- Research Scenario

  the test will be carried out in 2 different general scenarios with 3 sub-scenario, in full connected condition and a partial connected condition with five different scenarios.

  - Scenario 1

  In this 1st test will conducted the full connected scenario. Nodes 1 and 2 are nodes that are the object of the assessment. Node 1 will contact node 2 for 30 seconds, which is done 30 times. The results of testing the average value of throughput, delay, jitter, and packet loss will appear in the Iperf software.

  - Scenario 2

  Nodes 1 and 3 are nodes that are the object of the assessment. Node 1 will contact node 3 for 30 seconds and will be tested 30 times and performed 30 times. The test results of the average values of throughput, delay, jitter, and packet loss will appear in the Iperf software after.

  - Scenario 3

  Nodes 2 and 3 are nodes that are the object of the assessment. Node 2 will contact node 3 for 30 seconds, which is done 30 times. The results of testing the average value of using the values of throughput, delay, jitter, and packet loss will appear in the Iperf software.

  - Scenario 4

  In this fourth test scenario, the partially connected node scenario will be conducted. The same thing in scenario one will be done, but the difference is that there is an additional load by node 2. on nodes 1 and 2 will communicate through node 3 for 30 seconds and do 30 times. The results of testing the average value of using the results will appear in the iperf software.

  - Scenario 5

  In this 5th test scenario, the partially connected node scenario will be conducted. The same thing in the previous scenario will be done, but the difference is the additional load by node 3. Furthermore, nodes 1 and 3 will communicate through node 2 for 30 seconds and do 30 times. Testing the average value of the use of the results will appear in the iperf software.

  - Scenario 6

  In this fourth test scenario, the partially connected node scenario will be conducted. The same thing in the previous scenario will be done, but the difference is the additional load by node 3. Moreover, nodes 2 and 3 will communicate through node 1 for 30 seconds and do 30 times. Testing the average value of the use of the results will appear in the iperf software.

- System testing and parameters

  At this stage, testing of the system will be carried out. After the server and client are built, system testing is carried out with the following scenario:

  - Communication functionality between clients.
  - Quality of service (QoS) on VoIP service.
  - Setting the bandwidth limit that can be passed not more than 1 Mbps in order to get a consistent value [6].

The data collection process will be carried out and analyzed at this stage. the functionality between clients before testing the client-server must be registered through the client SIP. This research will conduct functionality testing with several parameters such as QoS on VoIP services, quality of using VoIP, throughput, jitter, delay, packet loss, and data obtained well after QOS testing on VoIP. Then from this test, it can be concluded that the performance of VoIP using Raspberry Pi can work well [8].

III. EXPERIMENT AND RESULT

This section consists of two sub sections, i.e. (i) Ping testing and (ii) Quality Of Service testing.

A. *Ping Testing*

- Scenario 1

The first scenario is testing the connectivity between node 1 and node 2 using the ping command. Connectivity testing ensures that the nodes are connected. The ping test was carried out by node 1 with an IP address of 192.168.0.5 and node 2 with an IP address of 192.168.0.6. Therefore, in the ping test, it can be stated that node 1 and node 2 are connected to each other.

- Scenario 2

The second scenario is testing connectivity between node 1 and node 3 using the ping command. The ping test was carried out by node 1 with an IP address of 192.168.0.5 and node 3 with an IP address of 192.168.0.9. Therefore, in the ping test, it can be stated that node 1 and node 3 are connected to each other.

- Scenario 3

The third scenario is testing the connectivity between node 2 and node 3 using the ping command. The ping test is carried out by node 2 with an IP address of 192.168.0.6 and node 3 with an IP address of 192.168.0.9. In the ping test, it can be stated that node 2 and node 3 are connected to each other.

- Scenario 4

The fourth scenario tests the connectivity between node 2 and node 3 through node 1 using the ping command. The ping test is carried out by node 2 with an IP address of 192.168.0.6 and node 3 with an IP address of 192.168.0.9. Through node 1 with the IP address 192.168.0.5 In the ping test, it can be stated that node 2 and node 3 through node 1 are connected to each other.

- Scenario 5

The fifth scenario tests the connectivity between node 1 and node 3 through node 2 using the ping command. The ping test is carried out by node 1 with an IP address of 192.168.0.5 and node 3 with an IP address of 192.168.0.9. Through node 3 with the IP address 192.168.0.6 In the ping test it can be stated that node 1 and node 3 through node 2 are connected to each other.

- Scenario 6

The sixth scenario tests the connectivity between node 1 and node 2 through node 3 using the ping command. Performed by node 2 with an IP address of 192.168.0.6 and node 3 with an IP address of 192.168.0.9. through node 3 with the IP address 192.168.0.5 In the ping test it can be stated that node 2 and node 3 through node 1 are connected to each other.

B. *Quality of Service Testing (QoS)*

- Throughput.

The throughput plot testing is carried out 30 times with 30 seconds for each test and uses iperf. In the throughput test, there are several sub-scenarios, with two nodes and three nodes scenario. The test results of the average throughput value can be compared with the average value set by the TIPHON standardization to determine the quality of the SOLSR routing protocol throughput on VoIP described in Table 1.

TABLE I. THROUGHPUT VALUES

Scenario	Throughput (Mbps)
Scenario 1	31.480
Scenario 2	31,500
Scenario 3	74,610
Scenario 4	31,490
Scenario 5	39,230
Scenario 6	139.870

Based on the results of the throughput value obtained, it shows the stability of the throughput value in the scenario that has been carried out. Therefore, it can be concluded that the use of the SOLSR routing protocol on manet networks using VoIP for throughput parameters is classified as very good according to the standards set by TIPHON with a value range of 76 to 100 Mbps.

- Packet Loss

Table 2 and Figure 3 show the comparison of packet loss values that occur very varied based on packet loss measurements carried out on the Manet network using the SOLSR routing protocol on VoIP. The average value of packet loss in the scenario of 2 nodes, 3 nodes, on the full connected test is 0.037%, 0%, and 0.35%. The partial connected method has a value of 0.10%, 0.27%, 1.38%. Based on the results obtained, the average value of packet loss shows that the resulting variation in the value of delay and jitter does not affect the average value of packet loss. The cause of the packet loss value has increased because it has exceeded the maximum limit of packets that can be received. The results of this average value can be concluded that using the SOLSR routing protocol on a manet network using VoIP is classified as a very good category according to the standards set by TIPHON with a value range of 0 to 2%.

TABLE II. PACKET LOSS VALUES

Scenario	Packet Loss (%)
Scenario 1	0,037
Scenario 2	0
Scenario 3	0,35
Scenario 4	0,10
Scenario 5	0,27
Scenario 6	1,38

- Jitter

Table 3 and Figure 4 shows results of the jitter values that occur based on the results of jitter measurements made on the Manet network using the SOLSR routing protocol on VoIP. The average value of jitter in the scenario of 2 nodes, 3 nodes. Sequentially, the full connected test scores 2.09ms, 2.09ms, and 4.08ms, and the partially connected test values 1.56ms, 5.43ms, 19.79ms. According to the results of the average jitter value obtained, it shows that the more nodes are used, the greater the average value of the resulting jitter. In this case, because the more nodes are used, the traversed paths will also increase. This results in the data transmission process taking a long time so that the resulting average jitter value is getting bigger. Based on the results of these average values, it can be concluded that the use of the SOLSR routing protocol on a manet network using VoIP is classified as a very good category according to the standards set by TIPHON with a value range of 1 to 75 ms.

TABLE III. JITTER VALUES

Scenario	Throughput (ms)
Scenario 1	2,09
Scenario 2	2,09
Scenario 3	4,08
Scenario 4	1,56
Scenario 5	5,43
Scenario 6	19,79

- Delay

Table 4 and Figure 5 show a graph matching the delay values that occur based on the results of delay measurements carried out on the manet network using the SOLSR routing protocol. The average value of delay in the 2 node, 3 node scenarios, has a value of 19.27 ms, 10.46 ms, and 6.06 ms full-connected scenarios. Moreover, testing on the partial connected has a value of 30.73 ms, 18.28 ms, and 20.26 ms. The results of the average delay value obtained show that the more nodes are used, the greater the resulting average delay value; this is because the more nodes are used, the paths that are traversed will also increase and result in the data transmission process taking a long time. Based on the results of these average values, it can be concluded that the use of the SOLSR routing protocol on a manet network using VoIP is classified as a very good category according to the standards set by TIPHON with a value range of 0 to 150ms.

TABLE IV. DELAY VALUES

Scenario	Throughput (ms)
Scenario 1	12,27
Scenario 2	10,46
Scenario 3	6,06
Scenario 4	30,73
Scenario 5	18,28
Scenario 6	20,26

IV. CONCLUSIONS

Based on the research results, the following conclusions can be drawn:

1.The VoIP test results on the SOLSR routing protocol show that the average value of the QoS delay parameter in the two nodes, three nodes, and four nodes, respectively, has a value of 18.09 ms, 24.99 ms, and 34.79 ms. in a very good category according to the TIPHON standard.

2. Voip test results on the SOLSR routing protocol show that the average value of the QoS jitter parameter in the two nodes, three nodes using the full connected and partially connected test methods. Sequentially, it has a value of 2.09 ms, 2.09 ms, and 4.08 ms. The full connected test and the partially connected test values are 1.56 ms, 5.43 ms, and 19.79 ms. Moreover, the results on these tests are classified into good categories according to the TIPHON standard.

3. Voip test results on the SOLSR routing protocol show that the value of the QoS packet loss parameter in the two nodes, three nodes scenario uses the full connected and partially connected test methods. respectively has a value of 0.037%, 0%, and 0.35%. The full connected test and the partially connected test values were 0.1%, 0.27%, and 1.38%. Moreover, the results on these tests are classified into good categories according to the TIPHON standard.

4. Voip test results on the SOLSR QoS routing protocol, SOLSR routing protocol throughput parameter using VoIP shows network stability with an average value in the two node, three node scenario 3.75 Mbps.

5. Effective QoS testing is carried out within 1 - 2 hours to avoid overhead on the Raspberry Pi device and a time lag of 20-30 minutes to avoid network congestion in each test.

REFERENCES

V. REFERENCES

[1] R. Putratama, "Kilas Balik 2019: Kejadiab Bencana Terkait Cuaca Iklim, dan Gempabumi, BMKG," http//www.bmkg.go.id, 2019.

[2] BNPB, "Data Informasi Bencana Indonesia," BNPB, 2020.

[3] R. P. Foundation, "Raspberry Pi 4 Model B Spesification," https://www.raspberrypi.org, 2020.

[4] C. System, CCNA Exploration 4.0 : Network Fundamental, 2009.

[5] S. Valentino, Evaluasi Performansi OLSR (Optimized Link State Routing) Pada, Bandung: Jurnal Teknik Komputer. Politeknik Telkom, 2010.

[6] S. Subandi, "Sistem Komunikasi Berbasis Wireles (VoIP) Menggunakan Raspberry Pi Pada Daerah Tak Terjangkau Sumber Daya Listrik," JINTEKNA, 2017.

[7] R.Handayani, "Voice Over Internet Protocol (VoIP) pada Jaringan Nirkabel berbasis Raspberry Pi," KINETIK, 2017.

[8] A. Margolang, "Analisis Perbandingan Protokol Better Approach To Mobile Ad Hoc Network (BATMAN) dengan Protokol BABEL untuk Layanan Voice Over Internet Protocol (VoIP) Pada Mobile Ad Hoc Network (MANET)," Universitas Sumatera Utara (USU), 2013.