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Performance of VoIP Using Routing Open Shortest Path First with Multi-Protocol Label Switching

Danang Sunandar[a], Abdi Wahab*[b], Mudrik Alaydrus[a]

[a]*Department of Electrical Engineering, Faculty of Engineering, Universitas Mercu Buana, Jakarta, Indonesia*
[b]*Department of Information System, Faculty of Computer Science, Universitas Mercu Buana, Jakarta, Indonesia*

Abstract. Voice over internet protocol is a communication technology in the world of computer network that can be used for sending voice or video and data transmission over Internet Protocol in real time. VoIP network can be implemented with Asterisk applications as a server to a Private Automatic Branch eXchange applied in a Graphical Network Simulator 3. In this study, VoIP communication using routing OSPF within MPLS will be calculated, the QoS value collected based on impact performance under normal condition and not normal condition (link failure) different variant bandwidth in the network. The results from the simulation show that in normal condition and not normal condition there is average delay value with routing OSPF 4 ms and routing OSPF with MPLS 5 ms, the value of jitter max which same of 6 ms using varian bandwidth from 256 kbps and 512 kbps. All of the QoS parameters, such as delay, jitter and packet loss will be compared to standard ITU-T G.114. This research can be extended with addition of another measurement or another protocol.

Keywords: QoS; VoIP; GNS3; OSPF; MPLS

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1. Introduction

The development and convergence of Internet and telecommunication, and also with their increased applications, dependent on the availability of large bandwidth, with its Quality of Service (QoS) arrangement require by network and elements inside it which provide full support for data security and network performance improvement. It requires data transmission technology that not only facilitates routing and discovering of the best path, but also provides security in data communication [1]. Currently, QoS is a very important warranty given to telecom users to improve services. Voice over internet protocol (VoIP) is the transport of communication services – especially for voice data - over network. However, there is a mixture between the common terms used in this area [2]. VoIP and Internet Telephony are also often used to describe this communication transport. According to the ITU, VoIP takes place based routing network in general, while VoIP is a subset of IP Telephony, it is for communication between device entirely or partially over the Internet. On its development, the telecommunication industries use the metro ethernet [3][4].

GNS3 will be used in this research especially for making the simulation and analyzing design QoS VoIP and data. The result value for simulations between voice and data with routing OSPF and MPLS with OSPF, will captured by Wireshark, and the parameters will be calculated are delay, jitter, throughput, and packet loss. All QoS parameters captured in this research will be compared with standard ITU-T G.114.

The simulation will be grouped into two group, the first is simulation under normal condition and the second is simulation under not normal condition. To achieve these two conditions technology used for this is Cisco IOS®, Cisco Modular QoS Command, provides a modular and highly extensible framework for deploying QoS, by standardizing the CLI and semantics for QoS features across all platforms that are supported by Cisco IOS Software [5][6].

*Corresponding author: abdi.wahab@mercubuana.ac.id
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This study uses only Cisco MQC-Line Interface Classification platform standard. The configuration is done on R1, R2, R3, R4, R5, R6, R7 and R8 routers. See the Fig. 1 and Fig. 2. That R7 and R8 are connected to PC and Servers VoIP that have Asterisk application installed function as PABX on this VoIP system.

Related research for this topic of study, there is [7] that implement various queuing mechanisms to improve Quality of Service (QoS) for VoIP over IPv4 and MPLS network. Another research, [8], discussing about QoS of VoIP in converged MPLS networks. And in [9], the paper proposes a mapping between Label Switch Path of MPLS and Access Category of IEEE802.11e wireless.

1.1 Open Shortest Path First (OSPF)

OSPF is a link state Internet routing protocol use a “hello protocol” for monitoring the link inside protocols near to other routers and to test the status of their links to their active link. The other line will be determined as a backup path by OSPF. The specifications of OSPF protocol are open to the public, and the OSPF routing algorithm is based on shortest path first [10]. OSPF is dynamic routing protocols that can guard, manage, and distribute inter-network routing information follow every changes in network dynamically. OSPF included in the IGP (Interior Gateway Protocol) category that has a wider and more efficient link-state compare to other IGP protocol [11][12].

1.2 Multi-Protocol Label Switching (MPLS)

Nowadays, MPLS is one of flexible solution to address the problem in networking, such as network-speed, scalability, and quality of service. MPLS has appeared as a sophisticated solution to handle the bandwidth and service requirements for next generation internet protocol. Quality of Service can work to the management of network performance regarding to delay, bandwidth, packet loss, and jitter which is crucial for optimizing the performance of application [13]. As a control plane guided protocol, it has technology which combines the intelligence of routing with the high performance of switching. Thus, the control information needed for exchange must be ready before forwarding the first data packet. Label Distribution Protocol (LDP) is used by MPLS to establish label mapping in MPLS network domain to facilitate much control over the packet path by referencing the incoming labels to the label information base (LIB) [14].

The working principle of MPLS is to combine switching speed at layer 2 with routing capabilities and scalability at layer 3. The way it works is by inserting a label between the layer 2 and layer 3 headers on forwarded packets. Labels are generated by the Label-Switching Router which acts as a liaison between the MPLS network and external networks. The label contains information on the destination of the next node to which the packet should be sent. Then the packet is forwarded to the next node, at this node the packet label will be removed and given a new label containing the next destination. Packets are forwarded in paths called LSP (Label Switching Path) and LDP (Label Distribution Protocol) is a new protocol for label distribution that binds information to LSRs in an MPLS network. LDP is used to map FECs to labels, in turn creating LSPS.

1.3 Quality of Service (QoS)

Qualities of services (QoS) of network calculated in this research are throughput, packet loss ratio and delay. Network analysis is carried out to determine the quality of system performance to network performance in a real environment. The first parameter will be calculated is throughput. It calculates the number of packets send in the network for every second. The formula of throughput shows in (1) below.

$$\text{Throughput (kbps)} = \frac{\text{Number of Packets Sent}}{\text{Time Taken}} \tag{1}$$

Packet loss is the number of packets that fail to be delivered to the destination via the transmission medium. Packet Loss Ratio represent the ratio of the number of lost packets to the total number of sent packets [15]. The formula of packet loss ratio in this research shown in (2).

$$\text{Packet Loss Ratio (\%)} = \frac{\text{Number of Packets Sent} - \text{Number of Packets Received}}{\text{Number of Packets Sent}} \tag{2}$$

The delay or latency is the time it takes the data to travel from origin to destination. It can be formulated as follows:

$$\text{Delay (seconds)} = \text{Packets Receiving Time} - \text{Packets Delivery Time} \quad (3)$$

2. Research Design

Figure 1 the overall network simulation topology that in connect onboard LAN routers R1, R2, R3, R4, R5, R6, R7 and R8 with using OSPF routing configuration and switch2 connect asterisk server, serve as the interface PC1 to client PC2 using communication VoIP and data with GNS3.

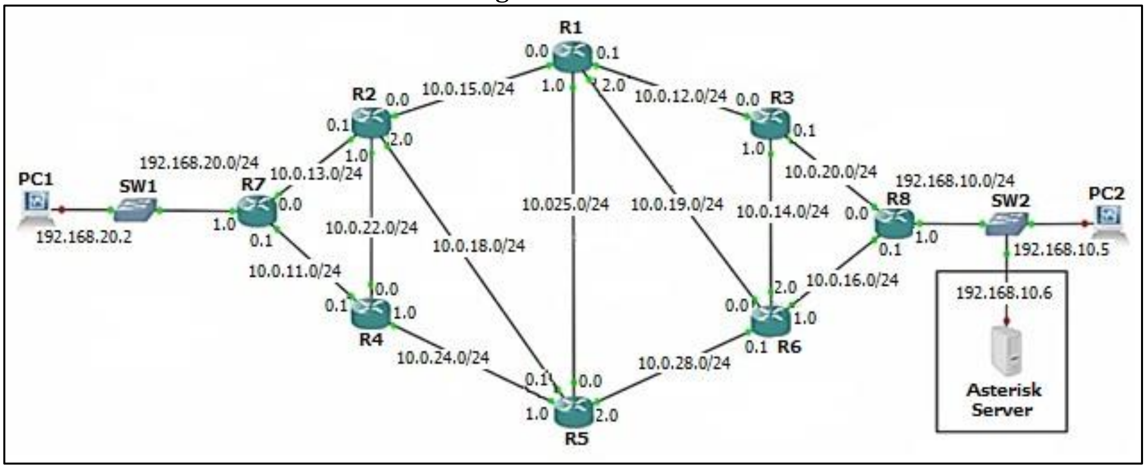


Figure 1. The overall topology to be implemented in simulation using GNS3 with OSPF routing

Figure 2 The overall network simulation topology that in connect onboard LAN routers R1, R2, R3, R4, R5, R6, R7, R8 and loopback router with using OSPF routing configuration and MPLS network that switch2 connect asterisk server, serve as the interface PC1 to client PC2 using communication VoIP and data with GNS3.

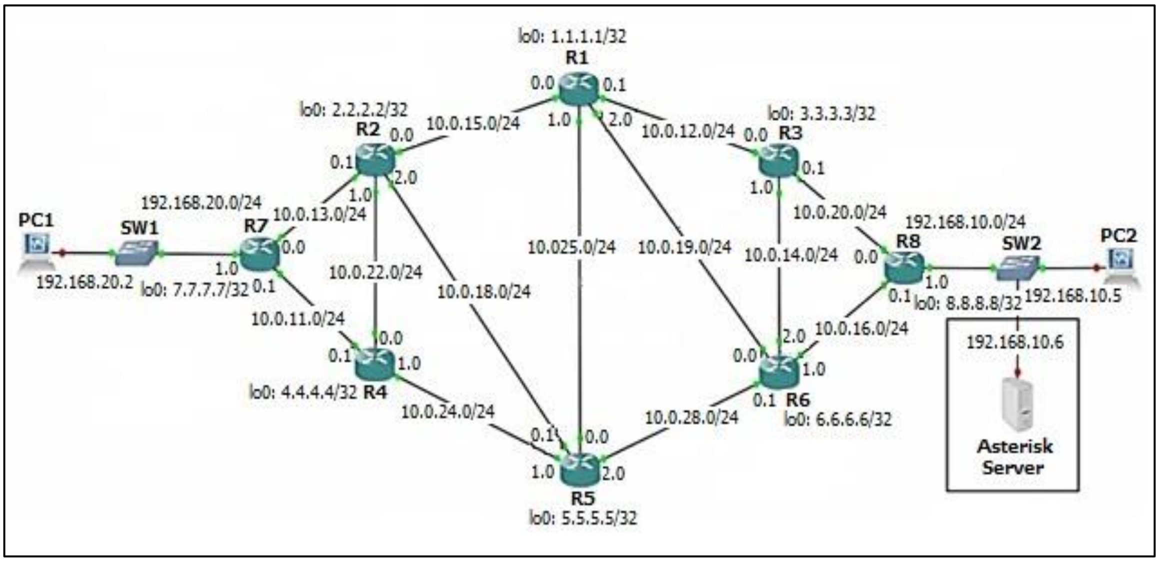


Figure 2. The overall topology to be implemented in simulation using GNS3 with Loopback, OSPF routing, and MPLS

In this research uses only Cisco MQC-Line Interface Classification platform standard. The configuration is router R1, R2, R3, R4, R5, R6, R7 and R8 routers. See the Fig. 1 and Fig. 2. That R7 and R8 are connected to PC and Servers VoIP that have Asterisk application installed function as PABX on this VoIP system. In this experiment requires 2 Pc and 8 routers where one of them is an Asterisk VoIP server included with GNS3 and FTP Server. The overall picture routing OSPF and MPLS, given Figure 1 and Figure 2, we try to take sample bandwidth 256 kbps and 512 kbps. Each bandwidth is performed six attempts with VoIP and data simulation transfer of data within two minutes for 40 Mb data.

This study measure Quality of Service (QoS) of the OSPF and OSPF with MPLS routing configuration, both of them will be perform in normal conditions and not normal conditions (link failure), where normal conditions of network routers is not link failure for all the routers. The not normal conditions there are 3 routers link failure, R1, R3 and R4. The change of bandwidth in the R7 and R8 that varied from 256 kbps and 512 kbps will be conducted. Data collection will be recorded using Wireshark with direct testing comparing the performance obtained from VoIP network with OSPF and OSPF with MPLS. Parameters of QoS extract from the collected data are delay, jitter, packet loss and throughput.

3. Result

3.1Measurement under normal condition with bandwidth 256 kbps

From the result normal conditions determine the bandwidth of the systems at 256 kbps, using the Wireshark application obtained average value results as in table 1, The result is considered very good by the standards of delay version ITU-T G.114 (4 ms/routing OSPF) and (6.1 ms/MPLS with OSPF routing). Result is considered very good by the standard of jitter version ITU-T G.114 (4.5 ms/routing OSPF) and category very good (6.7 ms/MPLS with OSPF routing). result is considered very good by the standards of packet loss version ITU-T G.114 (0%/routing OSPF) and (0%/MPLS with OSPF routing).

However, comparison average value of the measurement Quality of Service (QoS) as delay parameters standard ITU-T version TIPHON of the OSPF is 531.6 ms, category as bad according to delay standard ITU-T version TIPHON (delay range >450 ms). The average delay with MPLS is 175.2 ms, category as good according to delay standard ITU-T version TIPHON (delay range 150 up to 300 ms) [10].

Table 1. QoS for normal condition with bandwidth 256 kbps

Measurement	Delay (ms)	Jitter Max (ms)	Packet loss (%)	Throughput (bytes/s)
OSPF				
1	4	4.5	0	152675.4
2	4	4.8	0	140870.9
3	4	4.6	0	144559.3
4	4	4.7	0	142948.5
5	4	4.4	0	154512.6
6	4	4.5	0	150846.7
Average	4	4.5	0	147735.6
MPLS with OSPF				
1	6	6.7	0	80836.2
2	7	7	0	77714.1
3	6	6.7	0	81203.7
4	6	6.4	0	85190.3
5	6	6.8	0	81254.3
6	6	6.8	0	80383.1
Average	6	6.7	0	81096.5

3.2Measurement under normal condition with bandwidth 512 kbps

From the result normal conditions determine the bandwidth of the systems at 512 kbps, using the Wireshark application obtained average results as in table 2, The result is considered very good by the standards of delay version ITU-T G.114 (4 ms/routing OSPF) and (6,3 ms/MPLS with OSPF routing). Result is considered very good by the standard of jitter version ITU-T G.114 (4,5 ms/routing OSPF) and category very good (6,8 ms/MPLS with OSPF routing). result is considered very good by the standards of packet loss version ITU-T G.114 (0%/routing OSPF) and (0%/MPLS with OSPF routing).

However, comparison average value of the measurement Quality of Service (QoS) as delay parameters standard ITU-T version TIPHON of the OSPF is 113.2 ms, category as bad according to delay standard ITU-T version TIPHON (delay range >450 ms). The average delay with MPLS is 160.7

ms, category as good according to delay standard ITU-T version TIPHON (delay range 150 up to 300 ms) [10].

Table 2. QoS for normal condition with bandwidth 512 kbps

Measurement	Delay (ms)	Jitter Max (ms)	Packet loss (%)	Throughput (bytes/s)
OSPF				
1	4	4.6	0	146648.9
2	4	4.5	0	152114.2
3	4	4.7	0	144933.3
4	4	4.4	0	153838.7
5	4	4.6	0	147238.6
6	4	4.4	0	153838.7
Average	4	4.5	0	149768.7
MPLS with OSPF				
1	7	7.1	0	77421.2
2	6	6.9	0	78375.7
3	6	6.8	0	80346.7
4	6	6.7	0	80836.2
5	7	7	0	77714.1
6	6	6.7	0	81203.7
Average	6	6.8	0	79316.7

3.3Measurement under link failure condition with bandwidth 256 kbps

From the result not normal conditions determine the bandwidth of the system at 256 kbps, using the Wireshark application obtained average results as in Table 3 of link failure condition parameter. The result is considered very good by the standards of delay version ITU-T G.114 (5,8 ms/routing OSPF) and (5,8 ms/MPLS with OSPF routing). Result is considered very good by the standard of jitter version ITU-T G.114 (6,1 ms/routing OSPF) and category very good (6,5 ms/MPLS with OSPF routing). result is considered very good by the standards of packet loss version ITU-T G.114 (0%/routing OSPF) and (0%/MPLS with OSPF routing).

Table 3. QoS for not normal condition with bandwidth 256 kbps

Measurement	Delay (ms)	Jitter Max (ms)	Packet loss (%)	Throughput (bytes/s)
OSPF				
1	5	5.9	0	93333.7
2	6	6.3	0	87080.4
3	6	6.2	0	89913.1
4	6	6.1	0	90599.7
5	6	6	0	153052.4
6	6	6.2	0	89738.8
Average	5	6.1	0	100619.7
MPLS with OSPF				
1	5	6.7	0	81138.8
2	6	6.6	0	82518.5
3	6	6.7	0	81931.2
4	6	6.8	0	80339.2
5	6	6.3	0	86309.2
6	6	6.3	0	86148.7
Average	5	6.5	0	83064.2

3.4Measurement under link failure condition with bandwidth 512 kbps

From the result not normal conditions determine the bandwidth of the system at 512 kbps, using the Wireshark application obtained average results as in Table 4, of link failure condition parameter. The result is considered very good by the standards of delay version ITU-T G.114 (5,8 ms/routing OSPF) and (6,1 ms/MPLS with OSPF routing). Result is considered very good by the standard of jitter version ITU-T G.114 (6,1 ms/routing OSPF) and category very good (6,6 ms/MPLS with OSPF routing). result is considered very good by the standards of packet loss version ITU-T G.114 (0%/routing OSPF) and (0%/MPLS with OSPF routing).

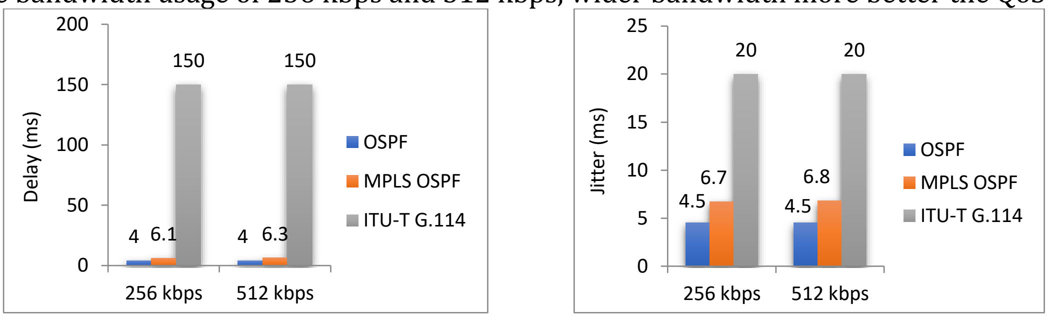
Table 4. QoS for not normal condition with bandwidth 512 kbps

Measurement	Delay (ms)	Jitter Max (ms)	Packet loss (%)	Throughput (bytes/s)
OSPF				
1	6	6.2	0	88112.6
2	5	5.8	0	94753.7
3	6	6.2	0	88307.1
4	6	6	0	94916.5
5	6	6.1	0	91423.2
6	6	6.3	0	86200.3
Average	5	6.1	0	90618.9
MPLS with OSPF				
1	6	6.8	0	79761.3
2	6	6.5	0	84867.6
3	7	7	0	78693.7
4	6	6.8	0	80339.2
5	6	6.3	0	86309.2
6	6	6.3	0	86148.7
Average	6	6.6	0	82686.6

3.5Discussion

Figure 3.a is the graph of delay under normal condition for different bandwidth 256 kbps and 512 kbps with routing OSPF, MPLS - OSPF and standard ITU-T. From the result normal conditions determine the bandwidth of 256 kbps and 512 kbps, using the Wireshark application obtained average value results as Figure 3.a, The result is bandwidth 256 kbps considered very good by the standards of delay version ITU-T G.114 (4 ms/routing OSPF) and (6,1 ms/MPLS with OSPF routing), result bandwidth 512 kbps is considered very good by the standards of delay version ITU-T G.114 (4 ms/routing OSPF) and (6,3 ms/MPLS with OSPF routing).

From this result we concluded that the addition of MPLS on normal conditions OSPF network did not improve the performance and efficiency for the network, it can see on the same delay value when using MPLS or not using it. Another factor effect same performance between using MPLS or not using it are the bandwidth usage of 256 kbps and 512 kbps, wider bandwidth more better the QoS value.



a b

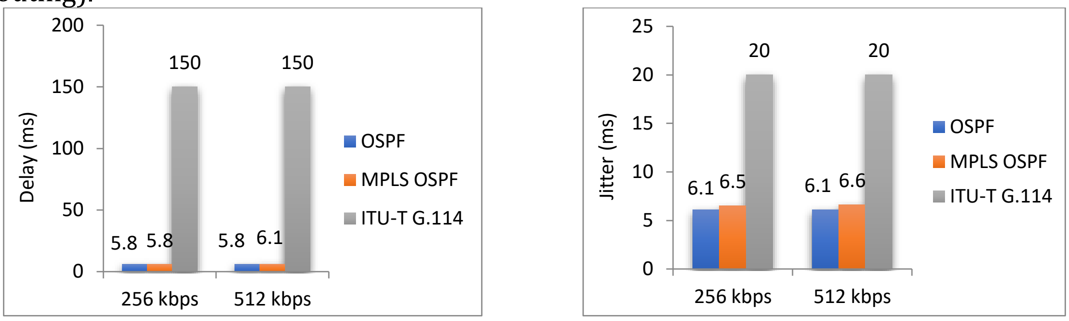
Figure 3. a. Delay (ms) under normal condition; b. Jitter (ms) under normal condition

Figure 3.b is the graph of jitter under normal condition for different bandwidth 256 kbps and 512 kbps with routing OSPF, MPLS - OSPF, and standard ITU-T. From the result normal conditions determine the bandwidth of 256 kbps and 512 kbps, using the wireshark application obtained average value results as Figure 3.b, The result is bandwidth 256 kbps considered very good by the standard of jitter version ITU-T G.114 (4.5 ms/routing OSPF) and category very good (6,7 ms/MPLS with OSPF routing), result bandwidth 512 kbps is considered very good by the standards of jitter version ITU-T G.114 (4,5 ms/routing OSPF) and (6,8 ms/MPLS with OSPF routing).

If we can see in the result, even the MPLS added in the normal condition OSPF network, the performance of network did not improve more efficient based on the jitter value. This condition is same when comparing the delay value and happens because need to be added big capacity of bandwidth in network MPLS.

Figure 4.a is the graph of delay under not normal conditions (link failure) for different bandwidth 256 kbps and 512 kbps with routing OSPF, MPLS - OSPF and standard ITU-T. From the result link failure conditions determine the bandwidth of 256 kbps and 512 kbps, using the Wireshark application obtained average value results as Figure 4.a, The result is bandwidth 256 kbps considered very good by the standards of delay version ITU-T G.114 (5,8 ms/routing OSPF) and (5,8 ms/MPLS with OSPF routing), result bandwidth 512 kbps is considered very good by the standards of delay version ITU-T G.114 (5,8 ms/routing OSPF) and (6,1 ms/MPLS with OSPF routing).

Figure 4.b is the graph of jitter under not normal conditions (link failure) for different bandwidth 256 kbps and 512 kbps with routing OSPF, MPLS - OSPF and standard ITU-T. From the result link failure conditions determine the bandwidth of 256 kbps and 512 kbps, using the Wireshark application obtained average value results as Figure 4.b, The result is bandwidth 256 kbps considered very good by the standard of jitter version ITU-T G.114 (6,1 ms/routing OSPF) and category very good (6,5 ms/MPLS with OSPF routing), result bandwidth 512 kbps is considered very good by the standards of jitter version ITU-T G.114 (6,1 ms/routing OSPF) and (6,6 ms/MPLS with OSPF routing).



a b

Figure 4. a. Delay (ms) under link failure condition; b. Jitter (ms) under link failure condition

Like in the previous result, the addition of MPLS in the not normal condition OSPF network did not give the performance improvement to the network. We can see in the delay and jitter value, there aren't any improvement in the QoS value. This condition occurs because the usage of bandwidth in 256 kbps and 512 kbps, more precisely because the delay and jitter value require large bandwidth for QoS.

4. Conclusion

Implementation QoS for VoIP is obtained using MPLS network protocol on each bandwidth, has a value of 1% is not much different from the value of routing OSPF in normal condition and not normal condition. The results, as written in the results table, show that, under normal condition and not normal condition affect the value of delay, jitter max and packet loss in network MPLS and Routing OSPF. The average value of delay for all of the scenarios are 6 ms. The bandwidth affect the value of jitter max advisable use more bandwidth for get the most, the measurement of the bandwidth 256 kbps and 512 kbps which same of 6 ms. In this measure VoIP and data is transmitted in its entirety in the absence of packet loss 0%, data transfer does not affect the VoIP and data based on this experiment. However, this process is an example study and can be considered to verify others languages as future work.

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