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Comparison of Voice over Internet Protocol (VoIP) Performances in Various Network Topologies

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ABSTRACT

VoIP is a digital communication technology that is currently developing because VoIP can be implemented on several network topologies, such as bus, star, and ring. Each of these topologies has advantages and disadvantages. So, a study is required to find out in which topology can VoIP be implemented optimally. In this research, VoIP is implemented in several topologies and furthermore the performance measurements are carried out for each topology. VQ manager is installed in order to measure the VoIP performance. For the server, we used Elastix and for the node implementation network topologies, we use several access points. From the results of the research, the performance of VoIP implemented in the star topology produces QoS that is better than other topologies with a delay value of 185 ms, 18 ms jitter, and 1% packet loss. This happens because in the star topology, all packets are distributed centrally. It is expected that the results of this research can be used as a reference in the application of VoIP technology in several types of topologies.

1. Introduction

VoIP is also known as IP Telephony which uses a computer network to transmit voice packet data from one place to another using an IP protocol intermediary. In other words, this technology is able to pass voice traffic in the form of packets over IP networks. It is the easiest way to make a phone call through the internet by sending packets through packet switched based network (Jalendry & Verma, 2015). VoIP grows due to the integration of VoIP system over the existing networking infrastructure (wireless) and low cost (Alharbi, Bahnasse, & Talea, 2017). VoIP is a technology that carries digital voice signals in the form of data packets with IP protocol. The incoming sound form is changed into a digital format. The process of sending sound data that has been converted into digital data can use the existing network infrastructure, so as to maximize the work process of the existing network.

VoIP itself is a technology that can be used for voice, video, and data communication using IP (Internet Protocol) and can be built with wired or wireless media (Wheeb, 2017). The use of VoIP-based telephone communications is very effective and efficient because this technology is more global and also cost-effective than traditional telephones. This is supported by the fact that communication with VoIP technology is not only limited to hardware, but can also use software such as x-lite or other VoIP softwares on Android. In computer networks, two or more nodes are connected together through a medium for the purpose of communicating data and sharing resources (Anjum & Pasha, 2015). There are some topologies in computer networks, such as bus, star ring, and mesh. In this paper, we focused on all of these topologies, except mesh topologies.

VoIP communication can run on different networks, so handover will occur. It must be taken into account that handover is the process of transferring Voice over IP call channels on an AP one Ethernet and then switching to the two Wireless Aps. In addition, the handover resulted in a loss of data. Therefore, pausing in conversation or data transfer is the best time for Handover. The handover process in embedded VoIP systems requires a long time so that it is able to meet capable Realtime service standards such as Voice Over Internet Protocol (VoIP).

The implementation of this interconnected wireless network certainly has advantages and disadvantages. One of the main advantages is wider coverage area, making it easier for users to connect to the network. However, the quality and performance of each nodes will be different because there is a factor of distance between the access points that causes delay.

This research is carried out to build a VoIP network with wireless technology that is implemented in different topologies and uses Elastix 5.0 as a VoIP server. Topology is a method to build a connection in networks. In previous research (Bisht & Singh, 2015), Each topology has been analyzed with different features. In this research, each topology has been tested with VoIP applications by take attention to the previous research (Alharbi, Bahnasse, & Talea, 2017). The purpose of this research is to obtain the optimum quality of VoIP in each topology. After the implementation is complete, QoS components are measured (such as delay, jitter, packet loss) on each nodes. This research is expected to be useful for the development of VoIP technology especially when implemented on a wireless network consisting of several nodes.

2. Literature review

2.1. Previous Research

Topology analysis was carried out earlier in a study (Bisht & Singh, 2015) and resulted in a detailed analysis of the advantages and disadvantages of each topology. Also resulted in similar analysis was research in VOIP implementation on VPN networks (Alharbi, Bahnasse, & Talea, 2017) using several VPN networks by producing results analysis. Structure analysis of each topology has also been conducted in a previous reserach (Kumar, Sherwani, & Singh, 2015), which resulted in analysis of each topology structure. A review about VoIP implementation has also been conducted in some research (Ayokunle, 2012; Wheeb, 2017). In their research, they focused on VoIP implementation done by using different server. Our goals in this research is to combine the research on network topologies and VoIP by paying attention to the previous research.

2.2. VoIP

VoIP technology is a new breakthrough in long distance communication through internet media. The process takes place by converting analog sound into digital data and sending it through computer networks. Currently, building a VoIP network is relatively easier compared to previous years. This is due to the large number of Open Source applications available for VoIP services. Figure 1 presents the implementation of VoIP.

One of the most influential factors to VoIP quality is codec-decoder (codec). Codec is an algorithm for compressing voice data that aims to reduce the number of bytes sent in the network. The use of the right codec in VoIP implementation is one of the decisive things in achieving the quality of VoIP communication [6]. The simplest form in the VoIP system is communication between two computers connected to the internet. Computers can enjoy VoIP services only by using a special software and supported with hardwares such as speakers and microphones. With the support of a special software, both computer users can be connected to each other in VoIP connections. Such connection can be in the form of exchanging files, sounds, or images. The main emphasis in VoIP is the relationship between the two in the form of sound.

2.3. Access Point

Wireless Access Point is a device that connects a wireless client (station) with a wired network (Wired LAN). Because it will connect two networks that use different media, the access point has the advantage of converting Ethernet frames into WLAN frames, and vice versa (Ayokunle, 2012). The Ethernet frame itself is a data format sent via a wired network, while the WLAN frame is a data format sent via wireless network (WLAN). An Access Point can convert 802.11 frames into 802.3 (Ethernet) frames or vice versa.

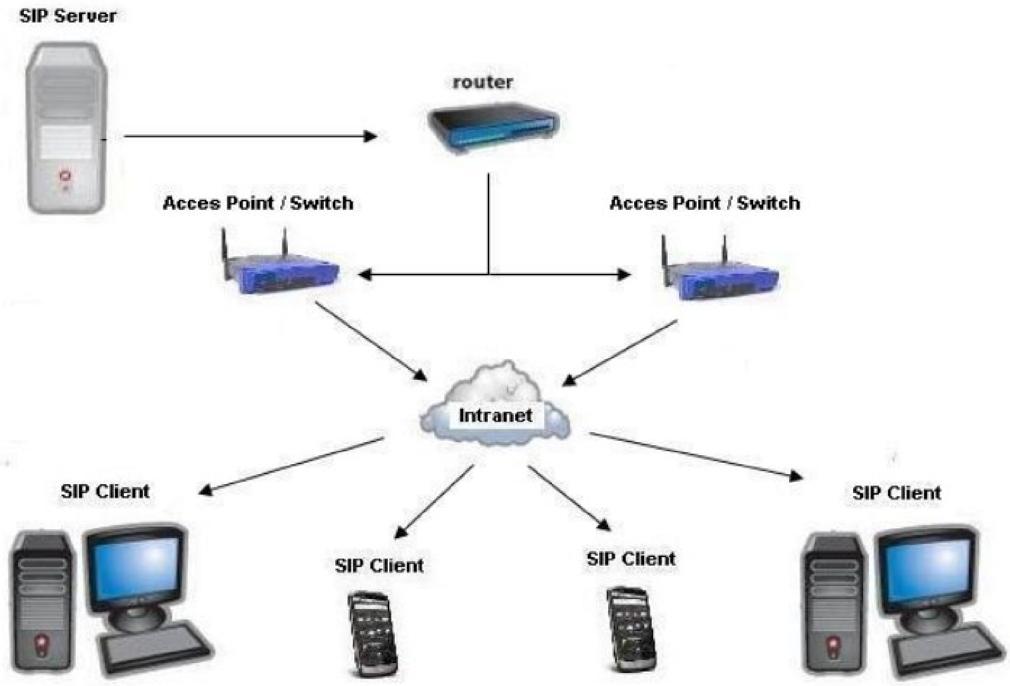

Figure 1. VoIP Implementation (Wibowo & Windarti, 2014)

2.4. Wireless Multi Nodes

Wireless multi nodes is a wireless network that is implemented with several interconnected nodes. The purpose of this method is to expand the range of an access point (G.Koshidgewar, 2015). Until now, two methods are available for creating such multi nodes network, namely repeaters and range extenders. These methods have similar function. Repeater or range extender functions as a repeater to expand signal range of the Wireless Router so as to reach clients locating far from access point.

Having similar function to strengthen the signal, this device comes with disadvantage compared to wireless repeats because the signals emitted will be displayed with different network names. So, if a user is out of range from the original signal coming out of an access point or router, the user must re-connect to the network with a different network name. With this function, a wireless repeater or wireless extender is urgently required. Both devices play an important role in internet network connection. The most significant difference is seen in the presentation of different network names when choosing to use a wireless extender.

2.5. Elastix

Elastix is an open source Unified Communications Server software that integrates IP PBX, email, IM, fax and collaboration functions. It has a web interface and includes capabilities such as call center software with predictive calls. Elastix functions are based on open source projects including Asterisk, Hylafax, Openfire, and Postfix. They offer PBX packages, fax, instant messaging, and email functions. Figure 2 presents the dashboard of Elastix.

Elastix developers have written web interfaces that allow you to access the program, making it appears like a complete product. Elastix also writes certain software such as program reporting, hardware detection, network configuration, software update modules, restore backups, and user management. Elastix was created by PaloSanto Solution, an Open Source based company support in Ecuador. Elastix was released to the public for the first time in March 2006. It was not a complete distribution but a Web interface for CDR (Call Detail Records).

Elastix Linux distribution is based on CentOS, which is a free and open source edition of Red Hat Enterprise Linux.

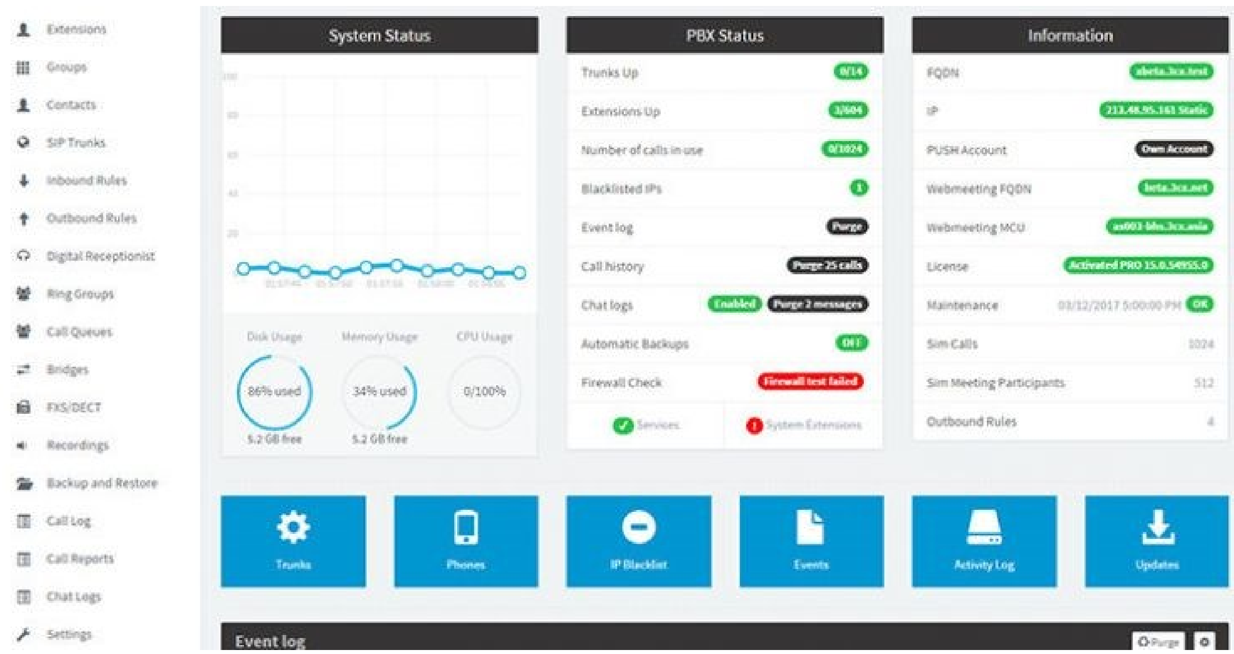

Figure 2. Elastix 5.0 Dashboard (Elastix, 2020)

2.6. QoS Parameters

The Quality of Service (QoS) and Capacity are two of the most important issues that require further study on wireless VoIP (Wheeb, 2017). In addition, several QoS criteria such as throughput, end-to-end delay, jitter, and packet loss has been considered to evaluate the performance of VoIP (Wheeb, 2017). The quality of VoIP services has several parameters that can be measured including:

a. Latency, delay which results in packet delay coming to the receiver. The maximum latency recommended by ITU for voice applications (VoIP) is 150 ms, but the maximum latency with sound quality that can still be received by users is 250 ms (Alharbi, Bahnasse, & Talea, 2017) .
b. Jitter, which is the difference in arrival time from the time the packet receives the expected time. It is defined as the delay variation between two successive packets in the same traffic flow (M. Sllame, 2017).
c. Packet loss, which is the loss of packets due to long queues or too large package size compared to the available bandwidth. It is defined as the number of lost packet during the communications between sender and receiver (M. Sllame, 2017).
d. Throughput, which is the total number of successful IP packet arrivals observed at the destination during a certain time interval divided by the duration of the time interval (equal to the number of successful IP packet shipments per service-second) (Alharbi, Bahnasse, & Talea, 2017).

3. Method

3.1. Specifications

The server specifications that we use in this study are explained in Table 1. We use Elastix 5.0 which has been bundled with IP PBX as a VoIP server. The installation process of Elastix 5.0 can be seen on the Elastix website. This version of VoIP server is very different from the previous one, because this version is supported by bundling from 3CX which is a world VoIP hardware and software manufacturer. The main features of Elastix 5.0 that do not exist in the previous version are:

a. ability to handle 16 concurrent calls.
   The number of 16 concurrent calls is very sufficient for the needs of VoIP services in an institution
b. supports use on a Mini PC/Raspberry.
   With supported use on mini PCs, VoIP services implementation becomes more flexible, energy efficient and in place.
c. Supports Google Cloud
   In order to enable this feature, application has to be installed. Whereas VoIP services can be used online without relying on the internet/local network that we implemented the server. With this feature, users can use VoIP services anywhere, provided they get an internet connection.

Table 1. VOIP Server Specifications

No	**Spesification**	**Description**
1	Prosesor	Core i3 2.2 GHz
2	RAM	8 GB
3	Hardisk	500 GB
4	LAN	Realtek PCIe Family Controller

Source: (Elastix, 2020)

3.2. Access Point & Router Specifications

Table 2 shows the access point specification that we use in the study. So, we can see that the AP used is standard wireless support with a maximum speed of 300 Mbps (standard n).

Table 2. Access Point Specificatons

No	Specificaton	Description
1	Wireless Standar	IEEE 802.11n, IEEE 802.11g, IEEE 802.11b
2	Interface	4x 10/100Mbps LAN PORTS 1x 10/100Mbps WAN PORT
3	Wireless Function	Enable/Disable Wireless Radio, WDS Bridge, WMM, Wireless Statistics

Source: (TP Link, 2020)

For the router implemented in ring topology, a mikrotik RB 850GX2 uses a router in order to connect both network in Access Point 2 and Access Point 3. Table 3 presents its specifications.

Table 3. Router Specificatons

No	Specificaton	Description
1	Ethernet	10/100/1000 5x Ethernet ports
2	Data Storage	512MB NAND memory chip, microSD slot
3	CPU	P1023 533MHz (Dual core)
4	Serial Port	One DB9 RS232C asynchronous serial port

Sourcer: (Mikrotik, 2020)

3.3. Network Implementations

The implementation of networks consists of different topologies, which are: bus, star, and ring. Each of the topology has its benefits and the drawbacks.

3.3.1. Bus Topology

The bus topology is the simplest topology in computer networks. It transmits message as the message arrives at each node, and the nodes checks the destination address contained in the message to see if it matches its own (Anjum & Pasha, 2015). If the address does not match, the node does nothing more. A major disadvantage of this network topology issued with security as data is broadcasted across the network and single point of failure if hub or backbone goes down (Kumar, Sherwani, & Singh, 2015). The most advantage of this topology is the cheapest from the all topologies implementation.

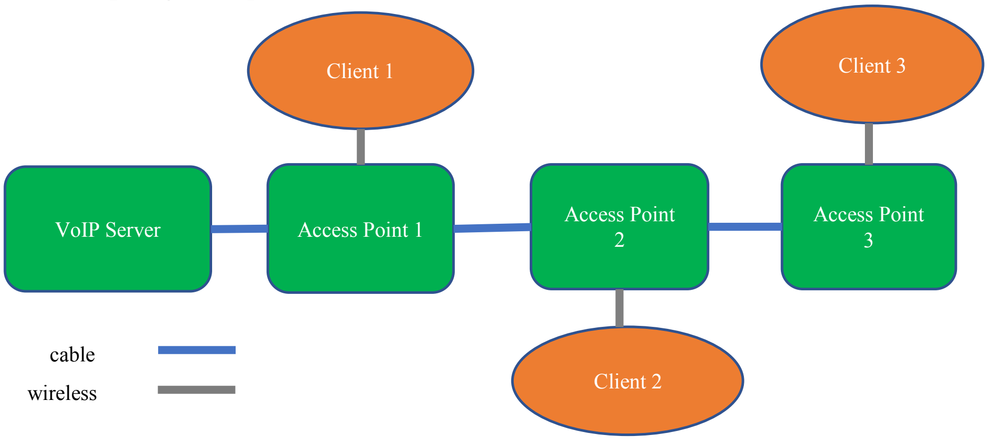

Figure 3. VoIP implementetaion Using Bus Topology

Figure 3 shows implementation of VoIP using bus topology. In this implementation, three access points are used as the nodes. Blue lines indicate connection by cable and grey lines mean by wireless.

3.3.2. Star Topology

In this topology, a central switch or hub is used to connect all the components (Bisht & Singh, 2015). The advantage of this topology is easy problem diagnose or maintenance. However, if the center of hub is down, the all communications will stop.

Figure 4 explains the implementation of VoIP using star topology. We can see that access point 1 are connected star to Access Point 2 and 3. As a result, Access Point I play as a role for the all communications.

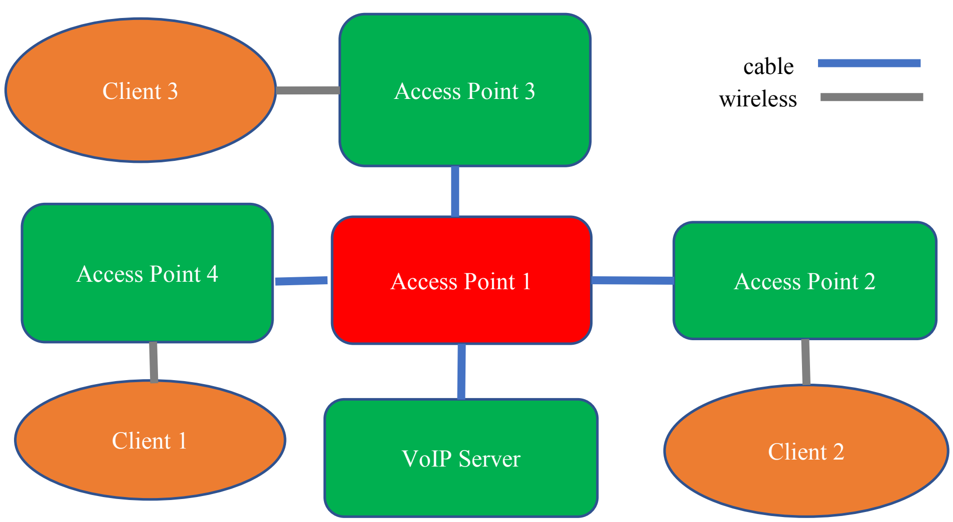

Figure 4. Implementation VoIP Using Star Topology

3.3.3. Ring Topology

In a ring topology, every device has exactly two neighbors for communication purposes (Singh & Ramola, 2014). All messages travel through a ring in the same direction (either "clockwise" or "counter clockwise") (Singh & Ramola, 2014). There is a direct point-to-point link between two neighboring nodes (the next and the previous nodes). These links are unidirectional which ensures that transmission by a node traverses the whole ring and comes back to the node (Singh & Ramola, 2014).

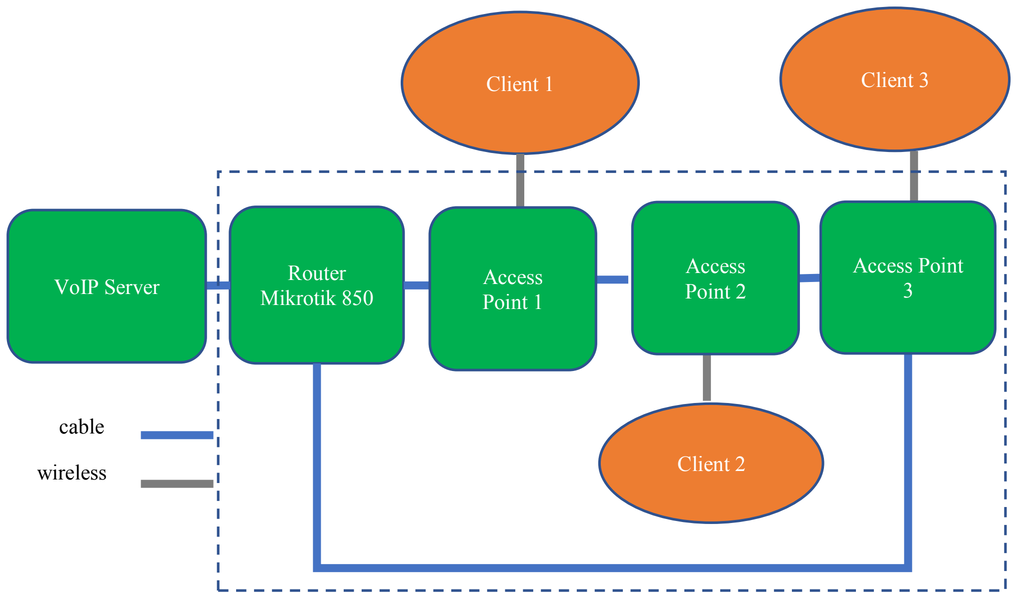

Figure 5. Implementation VoIP Using Ring Topology

In a ring topology, failure of one node on the network can affect the others. It is also difficult to add or remove nodes because it will change the network. The benefit of this topology is only the alternative routes. As explained earlier, the topology has two routes.

The implementation of VoIP using ring topology shown in Figure 5. Ring topology and bus topology are not actually distincly different. The difference only lies on the first node which is added a router in order to add two alternative routes to access point 1 and also a direct connection to access point 3.

3.4. Measurements

For performance measurements, we use VQ Manager as a program to determine the quality of VoIP. This program is installed on the server and is used to measure QoS quality (delay, jitter, packet loss) from ongoing communication. Following is the display of VQ manager when doing QoS data retrieval from ongoing communication. The result of this experiment is the average quality for two minute communications for each topology.

4. Result and Discussion

The results of this research focus on three primary QoS parameters, which are delay jitter and packet loss captured by Wireshark. In performing this study, the application is installed on each client. The data discussed in this section is obtained from the VoIP communications implemented by clients which are connected to access point 2 and access point 3.

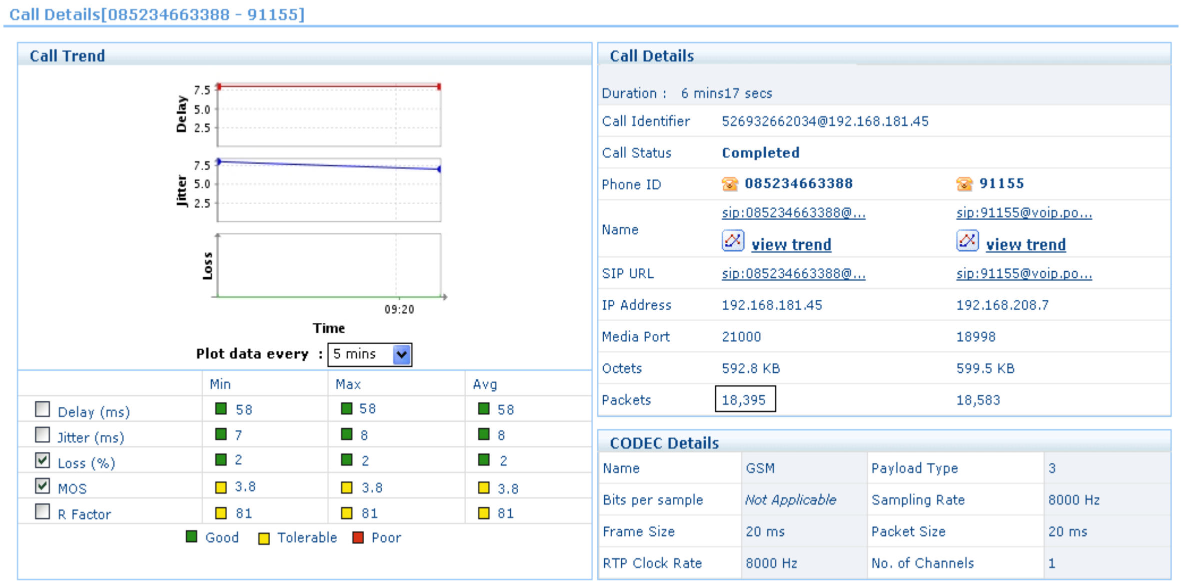

Figure 6. Measurement Using VQ Manager

4.1. Delay

In this experiment, delay means the total time needed for a VoIP packet to arrive at the destination. As we know, in this communication, packets will be sent first to the server, then forwarded to the recipient.

Figure 7 shows the average delay of VoIP communication in three different topologies. As expected, bus topology experienced the highest delay (around 240 ms) than the others. This happens because all nodes in this topology are connected in serial. As a result, a packet in the last node goes to server first by passing their neighbor. After the packet is received by the server, it will be sent to the receiver.

The ring topology also performed lower delay than the bus topology. It is because the ring topology has two alternative routes. In this topology, the packet will choose the shorter path to the server and the destination. The delay of star topology is the lowest from all topologies. It happens because the packet only passed maximum three nodes which are the center of access point, sender access point, and the destination. However, in other topologies, each packets should pass more than one nodes (access point) in order to facilitate communication because all of communication must be passed the server *(client – server - client)*.

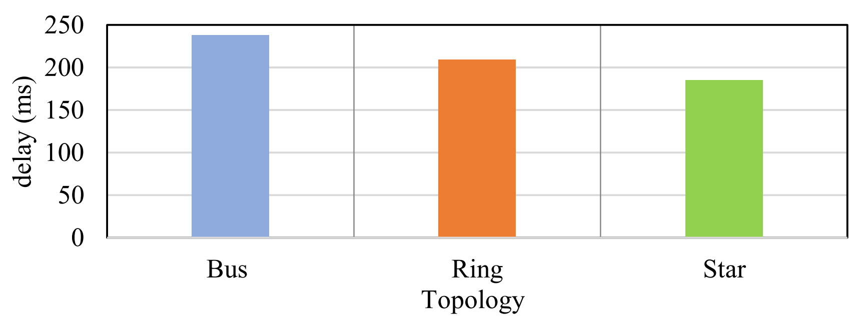

Figure 7. Average Delay of VoIP Communication in Different Topologies

4.2. Jitter

Jitter is a variation of delay packet between the previous package and the package afterwards. Jitter occurs because of a packet queue on the network. In this experiment, jitter depends on the number of nodes to communicate with clients and server.

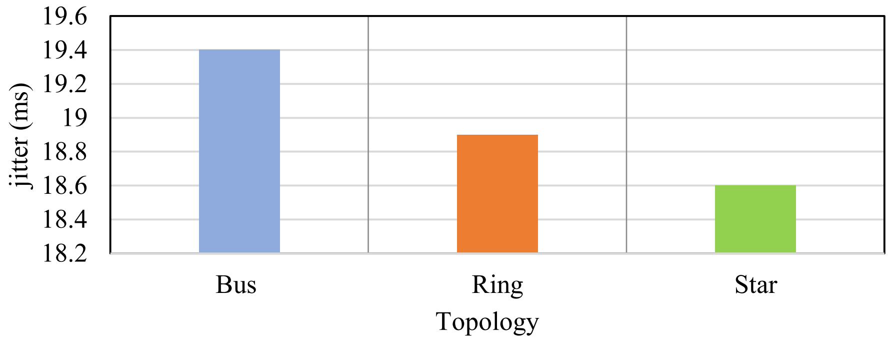

Figure 8. Average Jitter of VoIP Communication in Different Topologies

From Figure 8, we can see that the jitter of all topologies is relatively similar at around 18.5-19.5ms. This value is still acceptable for VoIP communication. The communication traffic is quite small because the communication taking place is only the one active communication that we performed. The jitter of bus topology is still the biggest due to more active nodes in facilitating communication between both clients and server. However, the star can direct transmit the data over the communication that resulting less variation of delay (jitter) in this topology.

4.3. Packet Loss

Packet loss here will be very influential from the large amount of traffic available. In the VoIP system, packet loss will appear because the type of protocol used is UDP. In addition, packet loss is caused by communication using wireless, which is vulnerable to noise.

Figure 9 shows the average of packet loss of VoIP communication in three different topologies. The packet loss of all topologies are quite similar with others and still acceptable around 1 %. From here, we can say that all topologies does not have much effect on the packet loss.

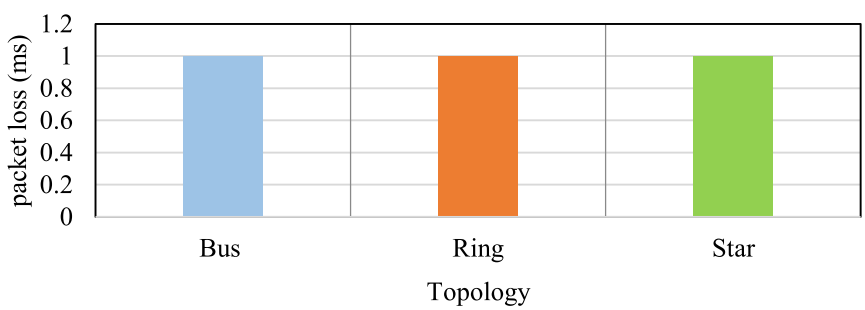

Figure 9. Average Packet Loss of VoIP Communication in Different Topologies

4.4. MOS

The factors determining voice performance include the MOS (Alharbi, Bahnasse, & Talea, 2017), varies from 1-5. This values are determined by the average of packet loss, delay and jitter. Figure 10 presents the average of MOS of VoIP communication in different topologies. From this measurement, we can see that the values of MOS of the three topologies are almost the same, which ranges between 4 - 4.1. It is because the values of delay, jitter, and packet loss of each topologies are still acceptable.

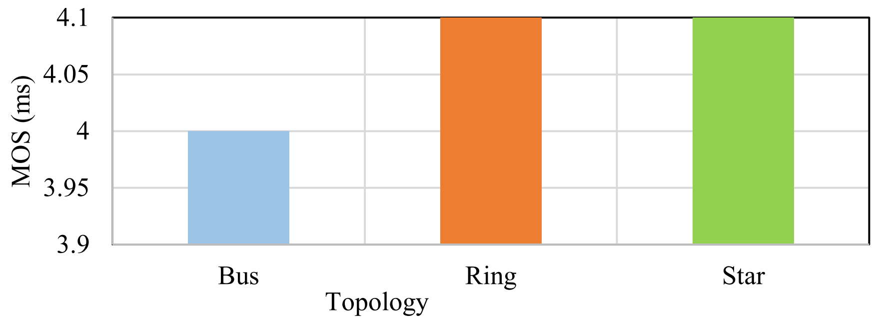

Figure 10. Average MOS of VoIP Communication in Different Topologies

5. Conclusion

We can conclude that each topology have drawbacks and advantages when implementing VoIP services. It is because each topology has different approach for communications. As the result, this affects the QoS, especially for the delay. From the experiments, it is found that star topology has the best QoS for VoIP, followed by ring topology, and then bus topology. In star topology, all communication data are directly transferred by the centre to the client or to the server. On the other hand, the ring and bus topology should send the data via the neighbour nodes that eventually affect the jitter. Therefore, the QoS gained are worse, especially for bus topology that do not have alternative routes. Finally, it is expected that future research would extend the experiment by taking into account the new interesting topology, namely mesh topology.

6. Acknowledgements

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