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The Impact of QoS Changes towards Network Performance

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ABSTRACT

Degrading or declining network performance in an installed computer network system is the most undesirable condition. There are several factors contribute to the decline of network performance, which their indications can be observed from quality changes in Quality of Service (QoS) parameters measurement result. This research proposes recommendations in improving network performance towards the changes of QoS parameters quality.

Keywords: *Network Performance, Quality Parameters, QoS Changes, Network Recommendation.*

1 INTRODUCTION

At the time talked about the Reliability of a system supporting network devices (network infrastructure). The main factors that influence it is Availability, Performance and Security, the relationship of these factors are as shown in Figure 1 [6].

Fig. 1, The formula for the achievement of reliability

Availability, availability of a service system must be maintained 24 hours straight without stopping to not be affected by the weather, working hours, holidays, leave employees, power outages, and so on, where the services and resources should be made available without interruption.

Performance, Performance of the service system is strongly influenced by the device used, from the core, distribution, access must be maintained not to occur failure and cause downtime for classical problems such as uncompatible, crashes, hangs, the lack of hardware support and technical services, and so on .

Security, useless expensive devices with guaranteed availability and performance is good but not safe, security and privacy issues regarding this very sensitive issue because there is no system that is safe while they are made by human hands, the system can be improved security built only from one level to another level .

Once a network is built, the next job more difficult it is to maintain the network still works as it should, in this case maintain network stability. If a device does not work then it will affect the work of the overall network. The network can be built with many technologies available. Although there are many reasons to monitor or manage the network, two main reasons are estimates for future changes and detect unexpected changes in the network. Unexpected changes can include such things as network devices that do not work, hackers trying to gain access to the network, or the fault lines of communication. Without the ability to monitor the network, the administrator can only react to problems when hackers appear even know after getting a report. Network management is the ability to control and monitor a computer network of a location.

2 QoS (QUALITY OF SERVICE)

QoS or in a free translation service quality is often called, is a mechanism that allows network applications or services can operate as expected. QoS can be defined as well as the ability to provide performance guarantee in the network. Performance is the speed and reliability of delivery of various types of load data in a communication system. Computer network performance may vary due to several problems, such as the problem of packet loss, delay (latency), jitter and throughput, which can make a big enough effect for many applications. For example, voice communications (such as IP Telephony or VoIP) and video streaming can make users frustrated when the application is streamed data packets over the network bandwidth is not enough, with a delay that cannot be predicted, or excessive jitter. Having regard to packet loss, delay (latency), jitter and throughput can be predicted and matched with the needs of the applications that are used in the existing network.

3 QoS PARAMETERS

QoS parameters that affect the performance of the network to be addressed here, as mentioned earlier is packet loss, delay (latency), jitter and throughput. Following exposure QoS parameters:

1. Packet Loss

Packet Loss is the failure of the transmission of IP packets to its destination. Packet loss is caused by a variety of possibilities, including:

- Congestion, caused because of excessive queues in the network
- Node, work exceeds the capacity of the buffer
- Memory is limited to nodes
- Policing or control of the network to ensure that the amount of traffic that flows to the amount of bandwidth, if the amount of traffic that flows in the network exceeds the bandwidth capacity, policing control will remove the excess of existing traffic.

Calculation to find the value of packet loss using the formula in equation (1) [source: Telkom Polytechnic].

$$\text{Packet loss} = \frac{\text{packets sent} - \text{packets received}}{\text{packets sent}} \text{ x } 100\% \quad \dots (1)$$

The standard value of packet loss can be seen in table 1. Packet loss occurs when packets are broken and discarded, or when the capacity of the network components exceeds the limit, which results in the packet is discarded. Packets can be damaged as they move across the wide area network, or when they traverse the network components such as routers and switches. This type of damage is detected in the process of "checksum". Checksum is the number of bits that are mathematically calculated by the sender and added to each packet. Recipients also calculate the checksum and comparing the calculated value with the value received by the package. If the received and the calculated checksum does not match, the receiver discards the packet. Regardless of the network topology, there is always a possibility that some level of packet loss can occur due to checksum to detect errors, mainly due to a large number of routers and switches traversed.

Table 1: Quality Standards TiPhone TR 101 329 for Packet Loss

	Category	Packet Loss
Packet Loss standard	Excellent	0 %
	Good	3 %
	Medium	15 %
	Poor	25 %

When it exceeds the capacity of the network components, congestion occurs on the component and the packet will be discarded. For example, if a packet arrives at a router at a rate faster than the router, the router can save them or send them, some number of packets will be discarded by the router. [2]. IP networks are now carrying heterogeneous mix of traffic, with different QoS requirements. Service models emerging multi-service packet networks, including packet backbones for 2.5G and 3G mobile networks, based on the ability of the network to guarantee QoS of user applications. End-to-end packet loss is one of the QoS performance metrics of the most significant, because the effect for many applications such VoIP, performance dropped dramatically if the packet loss exceeds a certain limit, and will become unusable if the packet loss is very large [3].

2. Delay (Latency)

Delay or Latency is the time delay caused by the transmission from one point to another point which becomes the goal. Delay in TCP/IP networks can be classified as follows:

- *Packetization Delay*
  Delay caused by the time required for the process of the formation of the IP packet of information users. This delay only occurs once, namely in resources.
- *Queuing Delay*

This delay is caused by the processing time required by the routers in handling packet transmission queue along the network.

- *Delay Propogasi*
  Delay Propogasi is in the process of traveling information during transmission media, such as SDH, coax or copper, which causes the delay, which is called the propagation delay.
- *Transmission Delay*
  Transmission Delay is the time it takes a packet to traverse a medium. Transmission delay is determined by the speed of the media and the size of the data packet.
- *Processing Delay*
  Processing Delay is the time required by a network device to see the route, change the header and task switching.

Calculation to find the value of the delay using the formula in equation (2) [source: Telkom Polytechnic].

$$\text{Delay} = \frac{\text{packet length (bit)}}{\text{link bandwith (bit/s)}} \text{ second ... (2)}$$

The standard delay value can be seen in Table 2.

Table 2: Quality Standards ITU-T G.114 for Delay

	Category	Delay
Delay (Latency) standard	Good	0 - 150 ms
	Medium	150 - 400 ms
	Poor	> 400 ms

To changes in the latency can be caused from the quality of the network components (cable / router / switch), serialization delay, routing and switching latencies, and queuing and buffer management. [5]
The quality of network components are factors that affect the propagation delay. Propagation delay is the time required information / data to travel at the speed of light in a medium of communication from the source to the destination. In free space, the speed of light is approximately 3 x 10^8 m/sec. The speed of light is lower in other media such as copper wire or fiber optic cable. Speed reduction caused by the type of transmission is called velocity factor (VF). Copper wire cable and fiber optic cable has a factor of nearly the same speed. Speed fiber-optic cable is usually about 70% of the speed of light while the copper wires varies from 40% to 80% depending on the construction. Coaxial cables are used many types have VF 66%. Satellite communication links using electromagnetic waves to propagate through the atmosphere and space information space. This information is converted from an electrical signal into a radio signal by the transmitter and the antenna. After this radio signals pass through the antenna, radio signals travel at the speed of light to the space in the room. As an example of the differences in the various media network, has done experiments in the process of sending an email from New York to London. It is assumed in the experiments conducted, the user is the only user is no channel of communication and distance from New York to London is 5458 km.
Here is an example of the calculation of the propagation delay = distance / speed;

- Email is sent using a copper link: 5458 / 197863,022 = 23.58 ms
- Email is sent using a fiber-optic link: 5458 / 209854.720 = 26.01 ms
- Email is sent using a radio link: 5458 / 299792,458 = 18.21 ms

Experiments conducted show the latency caused by the propagation delay in the transmission medium. Although the trial was a single user friendly and have unlimited bandwidth, packet rate will still be delayed by the propagation delay. This delay occurs regardless of the amount of transmitted data, the transmission rate and the protocol used.
Serialization is the conversion of a byte (8 bits) of data stored in the computer's memory into a serial bit stream to be transmitted over the communication medium. Serialization take a limited amount of time and is calculated as follows:
Serialization delay = packet size in bits / transmission rate in bits per second.
example:

- Serialization of 1500 byte packet using a 56K modem link is 214 milliseconds
- Serialization of 1500 byte packet using a 100 Mbps LAN is 120 microseconds

In IP (Internet Protocol) network such as the Internet, the IP packet is forwarded from the source to the destination through a series of IP routers or switches are continuously updating decisions about the best next router to get the packet to its destination. A router in a data transmission path can change the routing path that could ultimately affect latency. High performance IP routers and switches add about 200 microseconds of latency on the network due to packet processing. Assuming an average distance IP backbone routers is 800 km, 200 microseconds routing / switching delay is equivalent to the amount of latency caused by 40 km of fiber, routing / switching latency has contributed 5% of the end-to-end delay on the average internet network.

Another thing that happened in the transport layer is called "latency queuing". It refers to the amount of time an IP packet is spent in the queue while waiting for the utilization of excess transmission link out after routing / switching delay has been recorded. This can add up to 20 ms latency.

Latency network connection is the amount of time it takes the data to travel between sender and receiver. All computer networks have latency, the amount varies and can suddenly increase due to various reasons. Most people assume that the unexpected delay as lag. Latency on the Internet connection fluctuates in a small amount of one minute next time, but lag the addition of a small increase becomes more noticeable when running the application on-line.

3. Jitter

Jitter is the variation of the delay. Jitter is affected by variations in traffic load and the amount of collisions between packets (congestion) on the network. Jitter influence on network performance should be considered in conjunction delay. When large jitter delay is small but the performance of the network can not be said to be bad because of the amount of jitter can be compensated with a small delay value. Jitter will degrade the performance of the network when the value is great and also the value of delay is too large.

Calculation to find the value of jitter using the formula in equation (3) [source: Telkom Polytechnic].

$$Jitter = \frac{\sum variation\ delay}{\sum packet\ received}\ second \quad \ldots (3)$$

The standard value of jitter can be seen in table 3.

Table 3: Quality Standards ITU-T G.114 for Jitter

	Category	Jitter
Jitter standard	Good	0 s/d 20 ms
	Medium	20 s/d 50 ms
	Poor	> 50 ms

Jitter is generally caused by congestion in the IP network. Congestion can occur either at the interface of a router or network operator if the circuit is not set correctly. [7]

Congestion, the network context, refers to the state of the network in which the nodes or links carry so much data that can worsen the quality of network services, queuing delay, the data packet loss and blocking new connections. In a dense network, the response time is slow with a reduction in network throughput. Congetion occurs when insufficient bandwidth and network data traffic exceeds capacity. [4]

Jitter causes a packet to be delayed somewhere in the circuit, where there is no delay or queuing for other packages. This causes a variation in latency.

4. Throughput

Throughput is the actual bandwidth (actual) were measured in a particular time and in a certain network conditions that are used to transfer files of a certain size. System throughput is the sum of the speed of data that is sent to all terminals in a network.

Calculations in finding the throughput using the following equation as the formula in equation (4) [source: Telkom Polytechnic].

Standard throughput values can be seen in table 4.

$$\text{Throughput} = \frac{\sum \text{sent data (bit)}}{\text{time data delivery (s)}}\ \text{bps} \quad \ldots (4)$$

Table 4: Telkom Polytechnic Quality Standards for throughput

	Category	Throughput/Bandwidth
Throughput standard	Excellent	100 %
	Good	75 %
	Medium	50%
	Poor	< 25 %

Throughput despite having the same formula unit and the bandwidth, but the throughput is on describing the actual bandwidth at a certain time and on certain conditions and internet network that is used to download a file with a certain size. By simply using bandwidth as a benchmark, the file size of 64 kb should be downloaded in one second, but when measured turned out takes 4 seconds. So if the downloaded file size is 64 kb, while the download time is 4 seconds, then the actual bandwidth or can call throughput is 64 kb / 4 sec = 16 kbps. Some of the factors that determines the bandwidth and throughput are: network devices, network topology, the number of network users, electric induction and weather. [1]

4 PERFORMANCE RECOMMENDATIONS

Performance recommendations proposed in this study can be described as follows:

- Recommendations that can be done to overcome the adverse impact of network latency is a component can be in the review, while to overcome the latency queuing can be checked on the proxy server.

- Recommendations that can be done to overcome the adverse effects of jitter is to overcome the congestion caused by the lack of bandwidth and network traffic that exceeds the capacity of the bandwidth management can be used so that the bandwidth can be shared as needed, or can also be increased bandwidth.
- Recommendations that can be done to overcome the adverse impact of packet loss is network components can be in the review so that it can be seen whether there is a network component that is problematic and lead to packet loss. Meanwhile, to overcome congestion, as well as in the case of jitter, bandwidth management can be used so that the bandwidth can be shared as needed, or can also be increased bandwidth.
- Recommendations that can be done to overcome the adverse impact of network throughput is the component can be in the review so that it can be known whether there are problems of network components and the effect on throughput as the ability of a router / switch. Meanwhile, to overcome the problems caused by having multiple user throughput network, bandwidth management can be used so that the bandwidth can be shared as needed, or can also be increased bandwidth. Any network topology can be reviewed or changed in order to get a network topology that corresponds to the type of network.

Of the recommendation in the event of changes in impairment QoS parameters that cause the quality of the service to be slow / bad based on the recommendations mentioned earlier, the implementation of the recommendations can be seen in Figure 2.

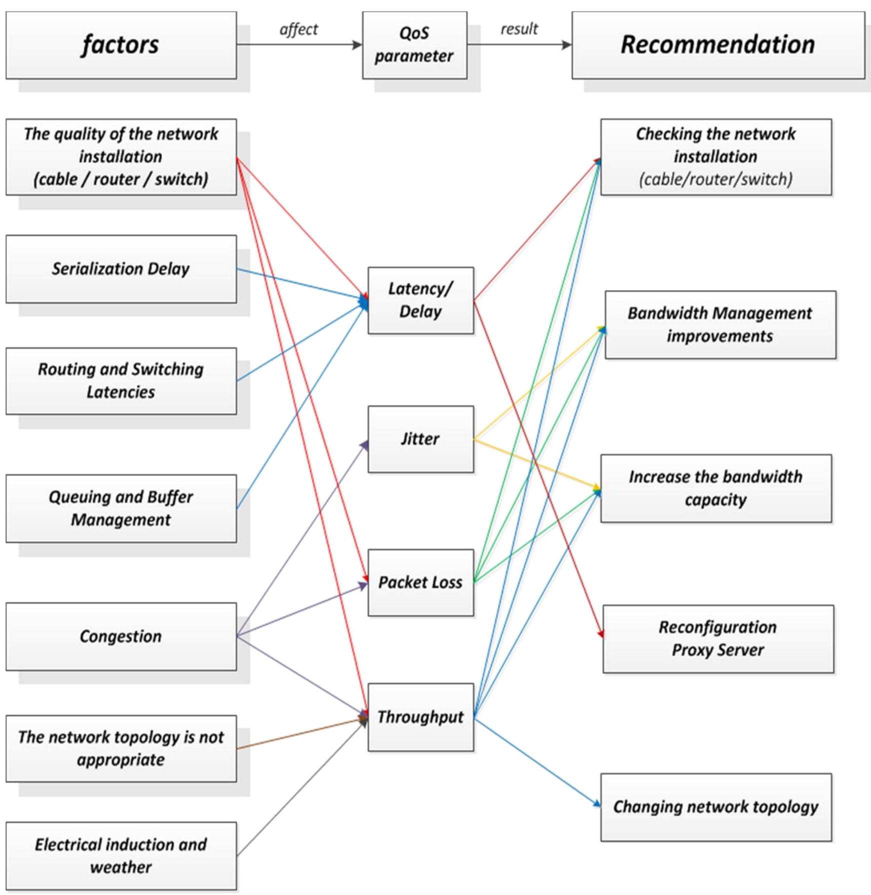

Fig. 2. Recommendations for QoS parameters

5 CONCLUSION

Modeling recommendation of the impact of changes in value of the QoS parameters resulting weakening of network performance. By keeping the quality of service in this case the quality of QoS is always monitored in order to obtain feedback as a recommendation maintain network performance factors, is expected to make a reference to maintain the performance of network attached fixed in prime condition.

6 REFERENCES

[1] Dewo.,E., S., Bandwidth dan Throughput, Artikel Populer IlmuKomputer.Com, 2003
[2] Grove., M., Impact of Packet Loss on BC/DR Deployments, Network Executive Software, Inc. (NetEx), 2010
[3] Hasib., M., Analysis of Packet Loss Probing in Packet Networks, Department of Electronic Engineering Queen Mary, University of London, 2006
[4] Janssen., C., Congestion, Technopedia (online), (http://www.techopedia.com/ definition/18506/congestion-networks), 2014
[5] Networks, 03b, What is Network Latency and Why Does It Matter?, o3bnetworks.com (online), (http://www.o3bnetworks.com/media /40980/white%20paper_latency%20matters.pdf), 2014
[6] Stiawan, D., Isu-isu utama pada Pengaturan Jaringan (Network Management), Jurusan Sistem Komputer FASILKOM UNSRI, 2009
[7] Support, Cisco., Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms), Document ID: 18902, Cisco Systems, Inc, 2006.