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Article

Comparative Analysis of Hierarchical Token Bucket and Per Connection Queue Methods in Video Conferences

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Abstract: Video conferencing is a set of interactive telecommunication technologies that allow two or more parties in different locations to interact using audio and video simultaneously. In video conferencing tools, bandwidth management is needed to maintain the quality of data transmitted through bandwidth. The Hierarchical Token Bucket (HTB) method is a method that uses a hierarchical structure and priorities for the client so that the distribution of bandwidth can be adjusted. In contrast, the Per Connection Queue (PCQ) method is a method that applies bandwidth sharing so that the allocation of bandwidth can be done more evenly to all clients. The parameters used to determine the quality of service in both methods are throughput, packet loss, delay, and jitter. The test results showed that in the Zoom application, the HTB method had an average TIPHON Standard Index of 3.5, while the PCQ method was 3.75. However, in the TrueConf application, the HTB method has a TIPHON standard index of 3.75, while the PCQ method has a TIPHON standard index of 3.5. In the TrueConf application, the HTB method is superior, while in the Zoom application, the PCQ method is superior.

Keywords: Hierarchy Token Bucket; Per-Connection Queue; Quality of Service; Video conference

1. Introduction

The development of telecommunications technology during the COVID-19 pandemic has increased very rapidly. Currently, many activities are carried out online using a video conferencing system. Video conferencing is a service used between two or more people to interact in the form of data, images, or sound in real-time. Now video conferencing has become a necessary facility for people affected by the COVID-19 pandemic [1][2][3][4].

Quality of Service (QoS) is a measurement of how good a network is and an attempt to define the characteristics and nature of a service. Several parameters are generally used in calculating the quality of service in a network, namely latency, jitter, packet loss, and throughput [5][6][7].

Some research is related to the comparative analysis of HTB and PCQ methods, one of which is the Implementation and Analysis of Hierarchical Token Bucket and Per Connection Queue Methods on Multi-Protocol Networks Label Switching Traffic Engineering for Voice over Internet Protocol Services. This study compares the HTB and PCQ methods for VoIP stabilization [8][9][10].

Hierarchical Token Bucket (HTB) is one approach used in network traffic management to improve quality and efficiency in video conferencing. In video conferencing, smooth, real-time audio and visual quality become very important to communicate effectively. HTB enables more sophisticated traffic management by setting priorities based on data type. This concept involves dividing traffic into different classes or priority levels. Each class has an allocated token quota, which decides when and how often data from that class will be transmitted over the network. By implementing the HTB method, video conferencing can take advantage of this feature by prioritizing critical video and audio data transmission for a better user experience. For example, participants' sound and video images in a video conference may have a higher priority than other data, such as web usage or background applications. This helps prevent quality degradation, audio or video lags, and data packet loss that can interfere with communication in video conferences. By using HTB in video conferencing, companies or organizations can ensure that network resources are used optimally while bandwidth usage remains by established priorities and policies. This positively impacts the attendee experience, improves the efficiency of virtual meetings, and ensures more effective and high-quality communication in an environment fraught with challenges and variations in network conditions [11][12][13].

For example, participants' sound and video images in a video conference may have a higher priority than other data, such as web usage or background applications. This helps prevent quality degradation, audio or video lags, and data packet loss that can interfere with communication in video conferences. By using HTB in video conferencing, companies or organizations can ensure that network resources are used optimally while bandwidth usage remains by established priorities and policies. This positively impacts the attendee experience, improves the efficiency of virtual meetings, and ensures more effective and high-quality communication in an environment fraught with challenges and variations in network conditions. PCQ can be used to ensure that each participant has an adequate bandwidth allocation according to their needs. This is important because different participants may have varying internet connection qualities, and PCQ can dynamically allocate resources to maintain video and audio quality. This way, participants with slower internet connections won't feel significant interference and can still participate comfortably [14][15][16].

In a video conferencing system, management is needed in distributing bandwidth, which aims to maintain bandwidth quality so that users can enjoy the network more effectively and efficiently. In addition to managing the needs of each user, it also regulates data traffic so that it continues to run smoothly. If there is no bandwidth regulation on a network, it will result in bandwidth control by one or several users. Bandwidth control is an activity carried out by users who use most of the available bandwidth on the network by downloading or streaming, thereby slowing down other user connections [17][18][19][20].

In general, problems can be solved using existing methods by balancing the traffic in the queue. In the PCQ method, it is done by grouping packets to distinguish one substream from another. In contrast to the settlement using the HTB method, which uses a fixed limit in implementation, if many users are active, HTB will choose the priority of the first active user if the bandwidth set per client is not divided evenly per user. However, each method has advantages and disadvantages for balancing traffic when conducting video conferencing. Based on the background described above, the authors intend to conduct further research on the comparative analysis of the HTB (Hierarchical Token Bucket) and PCQ (Per Connection Queue) methods.

2. Material and Method

2.1. Research Step

QoS (Quality of Service) is used in computer and telecommunication networks to manage, measure, and improve the quality of service provided to users. The main objective of quality of service is to ensure that network and communication services can provide consistent and reliable performance according to predetermined needs. Quality of service helps control and manage data flows, prioritize critical services, and allocate network resources wisely [21][22][23].

One crucial aspect of service quality is managing data packet queues in the network. With intelligent queue man-

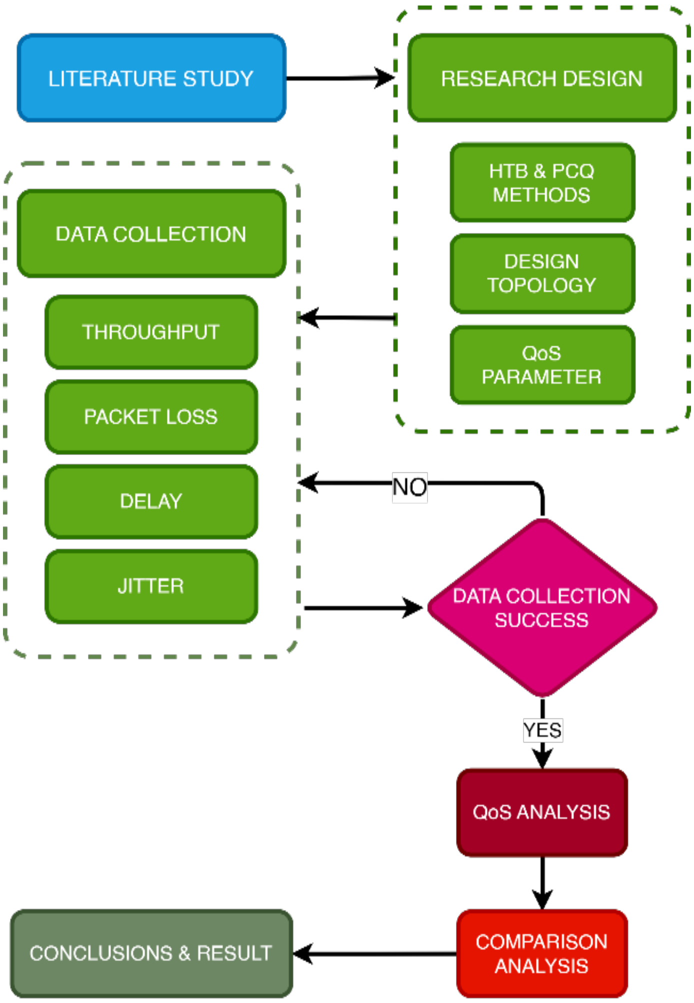

Figure 1. Stages of Research

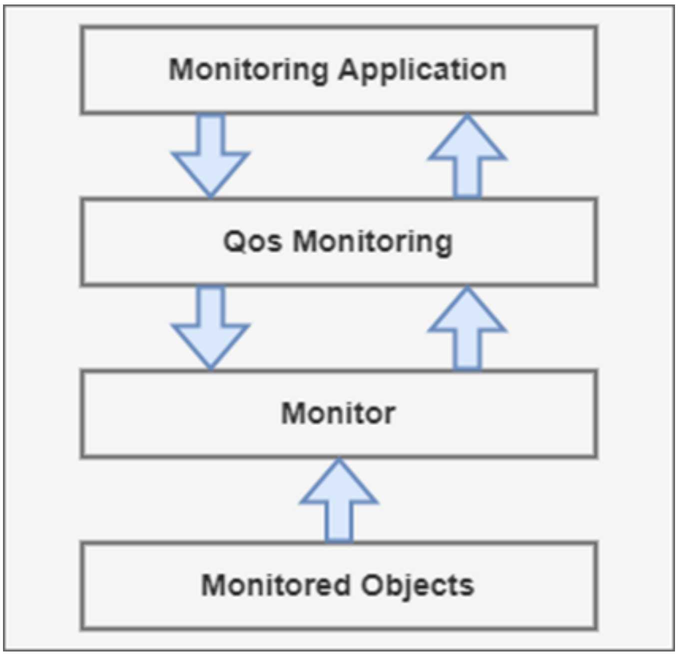

Figure 2. Model Monitoring QoS

agement, more critical data packets can be processed and sent ahead of less essential packets. The application of quality of service can provide benefits in various network environments, such as enterprise networks, telecommunication networks, or the Internet. By using quality of service, service providers can provide a better user experience, maintain the performance of critical applications, and ensure efficient use of network resources.

Quality of service can also control and reduce latency in the network, which is the time it takes for data to reach its destination. This latency reduction is significant for real-time applications such as voice and video calls that require instant response.

Predictable and scalable services are required in the context of campus networks because applications such as voice, video, and data are sensitive to latency across the network. In this test, we use quality of service to meet the traffic requirements of real-time sensitive applications such as voice and video and to prevent quality degradation due to packet loss, delay, and jitter [24][25].

The actions required in the data capture process are listed in Figure 1. Thus, implementing quality of service is a crucial step in ensuring that the campus network can operate efficiently, provide a quality user experience, and maintain the integrity of the data running on it.

Figure 1 shows the stages of video conferencing configuration in this study. It consists of several stages, which are explained as follows:

a) A literature study is carried out by studying supporting concepts and theories related to bandwidth management, video conferencing, and quality of service (QoS) needed in writing research through libraries related to research in the form of books and scientific journals.
b) This research builds a bandwidth management system in the network by using several pieces of hardware and software This research uses the hierarchical token bucket and per connection queue methods for conducting bandwidth management. The end of this study is to measure the scheduler's QoS (Quality of service). At the design stage of testing, the author will first perform a speed test on all clients to find out what the network speed is. clients in the network include 11 PCs and 1 mobile phone, some of which will conduct video conferences as activities carried out in testing. In handling data packets that need to be sent or received, bandwidth is one aspect that determines how much data can be received or sent simultaneously every second. Commonly used units are Mbps (Megabits per second) or Kbps (Kilobits per second). The comparative evaluation data of the two methods displayed are delay, throughput, packet loss, and jitter data, determining which method is better. The location of this research will be the MTI laboratory of Ahmad Dahlan University, Campus 3, Jln. Prof. Dr. Soepomo Sh., Warungboto, and Kec. Umbulharjo, Yogyakarta City.
c) At this stage, the author will collect data and information to be used as a reference in comparing two different methods based on QoS parameters (throughput, delay, packet loss, and jitter) to find a more optimal method of balancing traffic when conducting video conferences.
d) If successful in data collection, then the stage continues with data analysis, if not, then repeat data collection.
e) At this stage, the author begins to process and analyze the data or information that has been obtained at the data retrieval stage.
f) Take the final conclusion on the results of the bandwidth management comparison research on the video conference obtained and provide suggestions for input or criticism in order to improve future research.

Scenario testing in testing this system, proof will be carried out on the connectivity and reliability between the PCQ and HTB methods. The scenarios that will be carried out are:

a) Test connectivity without using bandwidth management: PC1, 2, 3, 4, and smartphones do video conference zoom with the camera position on, and PC5, 6, 7, 8, the camera position is off; PC9 downloads files of 4GB; PC10 watches on YouTube. PC1, 2, 3, 4, and smartphone perform TrueConf video conference with camera position on, PC5, 6, 7, 8, and PC9 with camera position off. Download a 4GB file on PC 10 and watch YouTube videos.

b) Testing the connectivity and bandwidth management of the PCQ method: PC1, 2, 3, 4, and smartphone do video conference zoom with camera position on and PC5, 6, 7, 8, camera position off, PC9 downloads a 4GB file, and PC10 watches YouTube.PC1, 2, 3, 4, and smartphone do TrueConf video conference with camera position on and PC5, 6, 7, 8, camera position off, PC9 downloads a 4GB file, and PC10 watches YouTube.

c) Testing the connectivity and bandwidth management of the PCQ method, PC1, 2, 3, 4, and smartphone do video conference zoom with camera position on and PC5, 6, 7, 8, camera position off, PC9 downloads a 4GB file, and PC10 watches YouTube. PC1, 2, 3, 4, and smartphone do TrueConf video conference with camera position on and PC5, 6, 7, 8, camera position off, PC9 downloads a 4GB file, and PC10 watches YouTube.

2.2. Equations QoS

An efficient enterprise network must provide predictable and scalable services when applications, such as delay-sensitive voice, video, and data, operate across the network. Organizations implement the concept of quality of service (Quality of Service) to meet the traffic requirements of sensitive applications, such as real-time voice and video, and to prevent quality degradation that may occur due to packet loss, delay, and jitter. Quality of service implementation allows organizations to control and manage data flows efficiently, prioritize critical services, and wisely allocate network resources.

One crucial aspect of Quality of service is the ability to manage the queuing of data packets in the network. With intelligent queue management, higher-criticality data packets can be processed and sent ahead of lower-criticality packets. Quality of service can be applied in various network environments, including enterprise networks, telecommunication networks, and Internet networks. By using quality of service, service providers can provide a better user experience, maintain the performance of critical applications, and ensure efficient use of network resources.

Controlling and reducing latency in the network is also an essential aspect of quality of service. Latency refers to the time taken by data to reach its destination. Latency reduction is crucial for real-time applications such as voice and video calls that require instant response from the network.

Predictable and scalable services are critical in the context of a campus network, and predictable and scalable services are critical as applications such as voice, video, and highly latency-sensitive data must operate across this network. In the testing phase, quality of service is used to meet the traffic requirements of applications that require real-time response, such as voice and video, while preventing quality degradation that can be caused by packet loss, delay, and jitter.

To implement quality of service, organizations must consider service level agreements (SLAs) with network service providers. These SLAs aim to guarantee a certain level of performance in the use of network services. It may include parameters such as latency, packet loss, and network availability that the service provider must adhere to.

Furthermore, in the context of quality-of-service model monitoring, monitoring the quality and performance of machine learning models is carried out continuously in a production environment. This aims to ensure that the model meets established quality standards and requirements. Quality of service model monitoring involves collecting and analyzing data to ensure that the model functions properly and produces the expected results.

The quality-of-service monitoring model consists of several essential components, including monitoring application components, quality of service monitoring, monitors, and monitored objects, all of which are listed in Figure 2. This entire monitoring system is designed to ensure that the machine learning model operates with a high level of quality and can cope with changes that may occur in the production environment [26][27][28][29].

The main purpose of the Figure 2 model above as QoS monitoring is to ensure that the machine learning model remains optimally performing, produces accurate predictions, and meets the quality requirements set according to the standard. By performing proper monitoring, you can identify problems quickly and take the necessary corrective actions to maintain model quality and performance on an ongoing basis, some of which are described in Figure 2 [30]:

a) Monitoring Application, Is an interface for network administrators. This component functions to retrieve data packet traffic information from the monitor, analyze it, and send the analysis results to the user. Based on the results of the analysis, a network administrator can perform other operations.

b) QoS Monitoring: Provides a QoS monitoring mechanism by retrieving information on QoS parameter values from data packet traffic.

c) Monitor, collect, and record data packet traffic information, which will then be sent to the monitoring application. The monitor measures the flow of data packets in real time and reports the results to the monitoring application.

d) Monitored objects are information such as attributes and activities monitored in the network. In the context of QoS monitoring, these are streams of data packets that are monitored in real time. The type of data packet flow can be known from the source and destination addresses in the IP layers, the ports used, such as UDP or TCP, and the parameters in RTP packets.

Some standards and formulas to determine the quality of a video with Quality-of-Service parameters consist of:

2.2.1. Throughput

Throughput is the speed (rate) of effective data transfer, which is measured in bps (bits per second). Throughput is the total number of successful packet arrivals observed at the destination during a given time interval divided by the duration of the time interval based on the standardization table in Table 1 as Equation 1.

$$Throughput\ \frac{Data\ Package\ Received}{Duration\ of\ observation} \times 100 \qquad (1)$$

Throughput is calculated through Equation 1 by dividing the amount of data successfully sent or received by the processing time. The amount of data is measured in bytes, while the data transmission time is measured in seconds. A throughput index value of 4 with a throughput more significant than 100 indicates excellent throughput quality. A value of 3 with a throughput between 75 indicates good throughput quality. A value of 2 with a throughput between 50 indicates adequate throughput quality. A value of 1 with a throughput between >25 indicates limited throughput quality.

Table 3 shows the throughput standard that measures how efficiently a system or process sends, receives, or processes data. This standard has four categories: excellent, good, medium, and bad. The good category is obtained if it is obtained very well if the throughput value is equal to 100; The good category is obtained if the Throughput value is 75; The category is obtained if the Throughput value is 50; And the bad category is obtained if the Throughput value is >25.

Table 1. Standardization Throughput.

Category	Throughput	Index
Very Good	100	4
Good	75	3
Medium	50	2
Bad	>25	1

Table 2. Standardization Delay.

Category	Big delay	Index
Very Good	<150 ms	4
Good	150 s/d 300ms	3
Medium	300 s/d 450 ms	2
Bad	>450 ms	1

2.2.2. Delay

Delay is the amount of time it takes for data to reach its destination. Delay can be influenced by distance and time, can be calculated using a predefined equation, and can be inferred based on the standardization table in Table 2. This aids in evaluating the quality of service and system performance in a more structured manner, enabling efficient monitoring and the identification of potential improvements needed in data delivery.

$$Delay(s)\,\frac{total\ delay}{total\ packet\ received} \times 100 \qquad (2)$$

Equation 2 calculates the average delay by dividing the total delay incurred in the system by the number of packets that have been successfully received. Total delay is the delay time in each packet sent, while the number of packets received is the number of packets that arrive.

Table 1 shows that if the Delay value is below <150 ms, the service shows excellent quality. The value range of 150 - 300 ms indicates that the quality of service remains good. If the Delay value is 300 - 450 ms, the service category shows a decrease in service quality. Meanwhile, if the Delay value exceeds >450 ms, the service shows low quality. The amount of standardization delay can be classified as shown in Table 2.

Table 2 shows the delay standards that serve as standardization for sending packages. This standard has four categories: very good, good, medium, and bad. The good category is obtained if very good is obtained if the delay value is <150 ms; the good category is obtained if the delay value is 150-300 ms; The category is obtained if the delay value is 300-450 ms; and the bad category is obtained if the delay value is >450 ms delay value is 300-450 ms; and the bad category is obtained if the delay value is >450 ms.

2.2.3. Jitter

Jitter is the difference between delays. Jitter can be calculated using a predefined equation, and can be inferred based on the standardization table in Table 3. This helps in understanding fluctuations in data delivery time,

Table 3. Standardization Jitter.

Category	Big Jitter	Index
Very Good	0 ms	4
Good	75 ms	3
Medium	125 ms	2
Bad	225 ms	1

Table 4. Standardization Packet loss.

Category	Packet loss	Index
Very Good	0%	4
Good	3%	3
Not Bad	15%	2
Bad	25%	1

which can impact the quality of service and overall system performance.

$$Jitter\frac{Data\ Package\ Received}{Duration\ of\ observation}\times 100 \quad (3)$$

Equation 3 shows the jitter from dividing the total delay variation by the total data packets received. At the same time, the total delay variation is obtained by subtracting the delay value from the average delay, as in Equation 2.

Jitter can be calculated using Equation 3, which includes variations in total Delay and total packet data received. The variation in Total Delay is calculated by subtracting the Delay on each data packet using Delay as in Equation 2. This explains the difference between the estimated arrival time of the package and the actual arrival time. The Jitter Index in Table 3 shows Index 4 values with 0ms jitter, indicating excellent network quality. Index 3 with a jitter of 75 ms indicates an acceptable variation in data transmission time. Index 2, with a jitter of 125 ms, shows a more significant variation in data transmission time. An index of 1 with 225 ms indicates unstable time variation and can interfere with real-time applications.

Table 3 shows the jitter standards standardizing network performance degradation. This standard has four categories: very good, good, medium, and bad. The good category is obtained if excellent is obtained if the jitter value is equal to 0 ms; The good category is obtained if the jitter value is 75 ms; The category is obtained if the jitter value is 125 ms; And the bad category is obtained if the jitter value 225 ms.

2.2.4. Packet loss

Packet loss is the number of packets that fail to reach their destination. Packet loss can be calculated by equation and can be concluded based on the standardization table in table 4 as shown in Equation 4.

$$PL=\frac{(Data\ Packets\ send-Data\ Package\ \ Receive\ )}{data\ packet\ send}\times 100 \quad (4)$$

Packet loss in Equation 4 involves subtracting the number of data packets received and the number of data packets sent, then dividing by the number of data packets sent, and multiplying by 100% to get a percentage. Table 4 shows the index values for the Packet loss value 0%, which indicates a very low packet loss rate and excellent service quality. Packet loss values of 3% indicate slight packet loss but are still considered good. The value of packet loss of 15% indicates a decrease in the quality of service and repairs that may be needed. Meanwhile, the value of Packet loss exceeding 25% indicates a serious network problem that significantly affects service quality and requires urgent corrective action.

The amount of standardization of packet loss can be classified as shown in Table 4. Table 4 shows packet loss standards that serve as conditions for the total number of packets lost. This standard has four categories: excellent, good, medium, and bad. The good category is obtained very well if the packet loss value is equal to 0%; The good category is obtained if the packet loss value is 3%; The category is obtained if the package loss value is 15%; And the bad category is obtained if the package loss value is 25%.

2.3. Tools and Materials

Wireshark is a network analysis software used to inspect, analyse, and record network traffic. While not a tool specifically used for video streaming, Wireshark can provide useful insights regarding video streaming in a network environment [31][32][33].

Wireshark can provide detailed information about network traffic during video streaming to check network performance. You can view bandwidth usage, latency, server response time, and other network metrics to understand how the network behaves when streaming video. In addition to the Quality of service (QoS) analysis, Wireshark can help measure and analyze service quality when streaming video. You can see QoS values related to video delivery, such as throughput, packet loss, and latency, which can help you evaluate the quality of your streaming experience [34]. The following tools and other materials used in this study are listed in Table 5.

2.4. Network Topology

Computer network topology is a technology that studies a technique to connect a computer with other computers and then form a network. Computer network topology is also a method to connect two or more computers

Table 5. Tools and Materials.

Tools and Materials	Specification
Personal Computer	Unit: 11 Processor: Core i7- 9700f RAM: 16GB Network Adapter: Intel Wireless-AC 9560 260MHz OS: Windows 11 Home Single Language
Xiaomi Redmi Not 9 Pro	Processor: Hisilicon kirin 659 RAM: 3GB Network Adapter: Wi-Fi 802.11 b/g/n
Access Point TP-Link	Model: TL-WA801ND
MikroTik Router-BOARD	Model: Rb450Gx4 GUI: WinBox
UTP Cables	Port: RJ-45

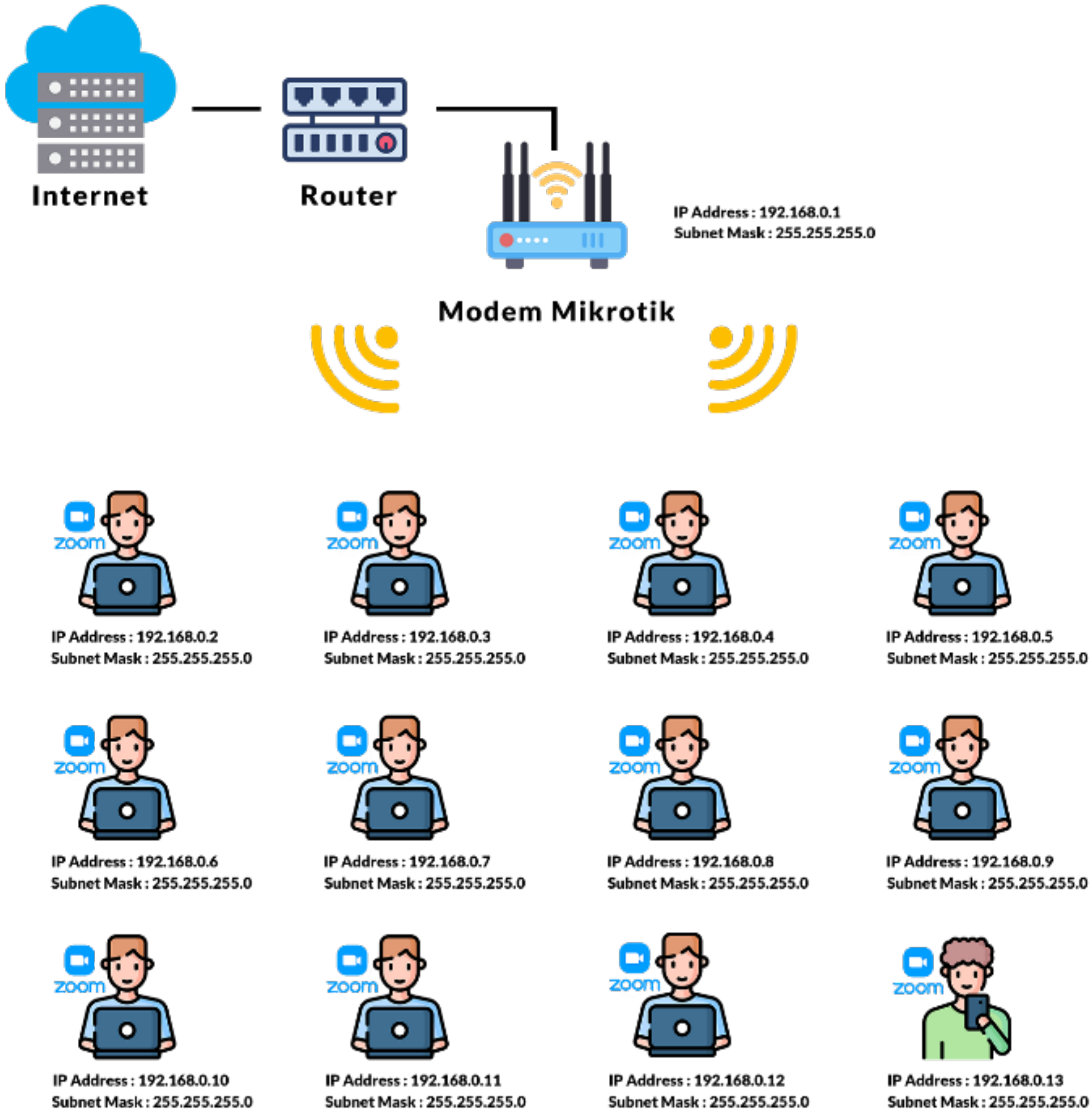

Figure 3. Network Topology

using UTP cables, fiber optics, or without wires (wireless) as a transmission medium [35][36]. In this case, it will be very possible for users to communicate with other users even though they are in different places The topology of a computer network will affect the speed of communication between computers. Basically, the basic topology MOF a computer network is a map of various computer networks. The network topology is divided into two parts: the physical topology and the logical topology [37][38][39].

The network topology in Figure 3 uses a star topology, where a router board is a gateway to connect to the internet. This router board is configured using the Per Connection Queue (PCQ) and Hierarchical Token Bucket (HTB) alternately to manage bandwidth, which functions as a control of internet usage.

In the PCQ method, network traffic is grouped by a specific connection or IP address. Each association or IP address will be given a specified priority or speed limit. In the configuration above, we create two types of queues named “pcq downstream” and “pcq_upstream” that implement the PCQ method. Each string has different speed limits for downstream (going to the user) and upstream (going to the internet) traffic flow. Next, we add two queue trees that direct traffic to the appropriate queues [40].

In the HTB method, network traffic is grouped into classes or groups with speed limits and their respective priorities. These classes form a hierarchy to organize traffic allocation and management in a structured manner. In the above configuration, we create two types of queues named “pcq default” and “pcq premium” that implement PCQ methods with different speed limits. Next, we add two queue trees that direct traffic to the appropriate queues.

Table 6. Test results on zoom.

QoS Parameters	No Bw Management		PCQ Method		HTB Method	
	Index	Category	Index	Category	Index	Category
Throughput	1	Ugly	3	Good	2	Currently
Delay	4	Very Good	4	Very Good	4	Very Good
Packet Loss	4	Very Good	4	Very Good	4	Very Good
Jitter	4	Very Good	4	Very Good	4	Very Good
Average	**3.25**	**Good**	**3.75**	**Good**	**3.5**	**Good**

Table 7. Test results on TrueConf.

QoS Parameters	No Bw Management		PCQ Method		HTB Method	
	Index	Category	Index	Category	Index	Category
Throughput	1	Ugly	2	Currently	3	Good
Delay	4	Very Good	4	Very Good	4	Very Good
Packet Loss	4	Very Good	4	Very Good	4	Very Good
Jitter	4	Very Good	4	Very Good	4	Very Good
Average	**3.25**	**Good**	**3.5**	**Good**	**3.75**	**Good**

3. Results and Discussion

The HTB (hierarchical token bucket) and PCQ (per connection queue) methods are two methods used in managing quality of service (QoS) on computer networks. Both of these methods aim to regulate the allocation and management of network traffic.

The TIPHON standard reference assessment is a method for measuring and comparing the performance of different technologies and solutions in a network environment. However, it should be noted that I need access to the results of current research or experiments that may have been conducted after my last knowledge in May 2023.

Refer to the latest research or literature on the TIPHON standard reference assessment to obtain more accurate information regarding the final comparison between HTB and PCQ methods. To get a good network, JDI must do a video conference scenario to determine the stability of network quality. After conducting several experimental techniques, the final comparison value between the HTB and PCQ methods was obtained using the TIPHON standard reference assessment, with results in the Table 6.

Based on the test results, the average index value in Table 6 shows that the method per connection queue is superior between the two methods, Hierarchical Token Bucket and Per Connection Queue at Zoom, based on the index. Within 11 minutes of throughput, packet loss, delay, and jitter all achieved good scores, with an average overall total index of No Bw Management 3,25 PCQ Method 3.75 and HTB Method 3.5. Hierarchical Token Bucket: he knows the performance between the two methods; the tested topology can provide high-quality service with good delivery speed, low packet loss rate, minimal delay, and stable jitter. To get the best video conference quality. with results per the Table 7.

Based on the test results, the average index value in Table 7 shows that between the two methods, Hierarchical Token Bucket and Per Connection Queue in TrueConf based on the index, the method per connection queue is superior. Within 11 minutes of throughput, packet loss, delay, and jitter all achieved good scores, with No Bw Management's overall average total index of No Bw Management 3.25, PCQ Method 3.5, and Htb Method 3.75. Hierarchical Token Bucket: it knows the performance between two methods. The tested topology can provide high-quality services with good delivery speed, low packet loss rate, minimal delay, and stable jitter. To get the best video conference quality.

In Figure 4, it can be observed that the throughput results when not using bandwidth management are only below 10% compared to the results when using the HTB video conferencing method. This indicates that the implementation of bandwidth management using the HTB video conferencing method provides a significant improvement in network throughput utilization, thereby optimizing the performance of video applications and ensuring a smoother user experience.

In Figure 5, it is evident that the highest packet loss results occur in both methods when utilizing the TrueConf video conferencing. This observation highlights that, irrespective of the network management methods employed, TrueConf video conferencing continues to encounter a noteworthy level of packet loss. Addressing this issue should be a priority to enhance the overall quality and reliability of the video conferencing experience.

In the chart in Figure 6, it can be seen that the latency in both methods is within safe limits and falls into the 'good' category according to TIPHON standards. This indicates that, in terms of latency, both methods meet the TIPHON standards well, implying that network quality in terms of latency can be considered good and can support

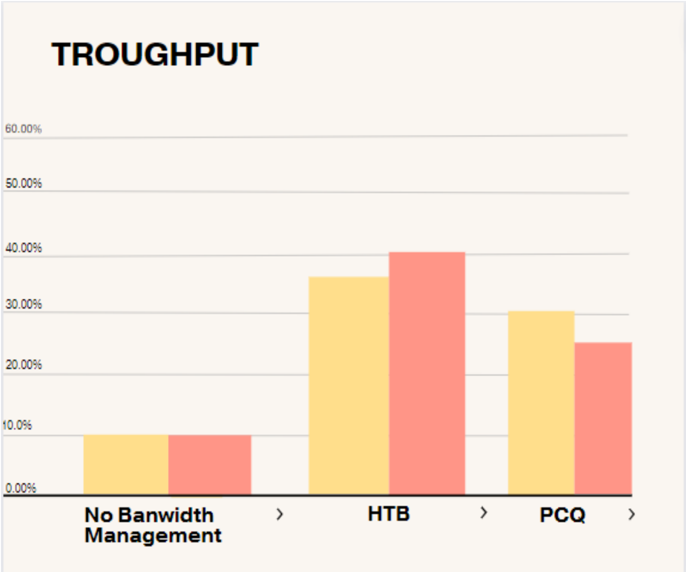

Figure 4. Throughput Graphic Comparison

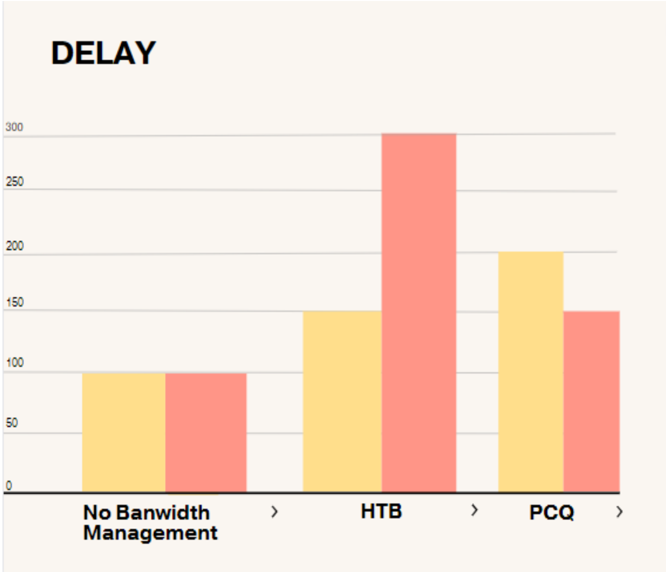

Figure 6. Delay Graphic Comparison

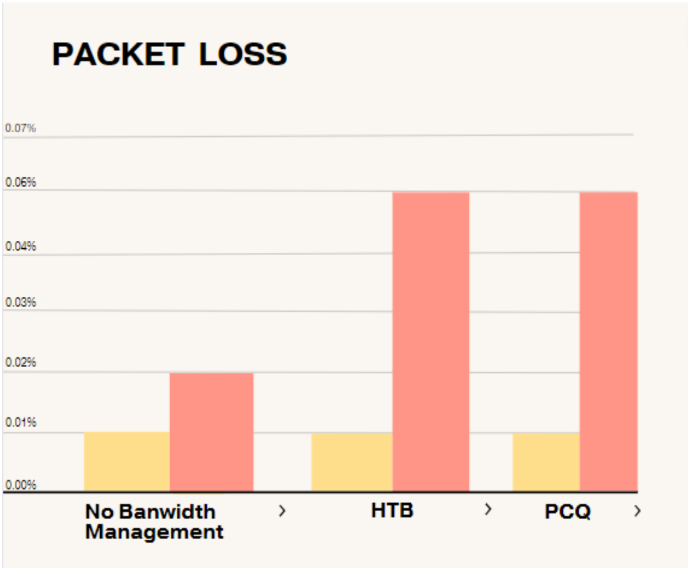

Figure 5. Packet Loss Graphic Comparison

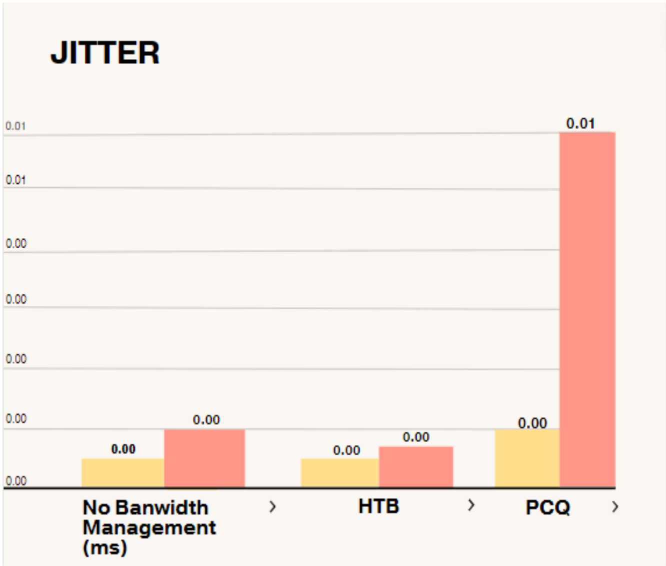

Figure 7. Jitter Graphic Comparison

time-sensitive applications such as real-time voice and video services.

In the chart in Figure 7, it can be seen that the jitter in both methods is in a safe condition and falls into the 'good' category according to TIPHON standards. This indicates that, in terms of jitter, both methods meet the TIPHON standards well, signifying that the fluctuation in data delivery time is relatively stable and within acceptable limits, thus supporting good communication quality, particularly for real-time voice and video applications.

As a result of comparing QoS values based on Tables 2 and 3, tests without using bandwidth management, we got an average value of 3.25 on Zoom and TrueConf. Using the HTB method, bandwidth management obtained an average index value of 3.5 with a good category on Zoom video conferences and 3.75 on TrueConf, also with a good type. While on PCQ, we obtained an average index value of 3.75 on Zoom with a good category and 3.5 on TrueConf with a good class. The values are similar after comparing the final values for these two methods. However, when compared again with each parameter in detail, each parameter will have differences.

Based on the data obtained with the Wireshark application and presented in the form of tables, The average value of throughput parameters without using bandwidth management was obtained at a value of 3.0% at Zoom and 2.6% at TrueConf video conferencing, both of which fell into the bad category. The HTB method got a percentage value of 44% when zooming with the "medium" type and 64% on TrueConf video conferences with the "good" category. In comparison, the throughput value in the PCQ method obtained a percentage value of 56% when zooming video conferences with the "good" type and 36% when doing TrueConf video conferences with the "medium" category.

The analysis results of the delay parameter values obtained without bandwidth management were 5 ms when zooming and 5 ms when TrueConf in the HTB method and

7 ms and 6 ms, respectively, when TrueConf. In comparison, the delay parameter values in the PCQ method were 5 ms at Zoom and 12 ms when TrueConf. The loss package on both ways and all test scenarios gets the same index value of 0% and falls into the "very good" category. The jitter value in all two methods and all test scenarios also gets the same index value of 0 ms, also in the "very good" category.

Some limitations of this study are that the internet network from switches in the IoT laboratory tends to be unstable. All clients in the test scenario use the same network, and the distance between all PCs and the access point is approximately the same. The data retrieval process is also carried out on one client PC, and testing is done only once in each scenario.

4. Conclusion

Based on the results of trials that have been carried out to compare two different bandwidth management methods, it was concluded that the two methods have differences that tend to be very thin when referring to the four parameters used as a comparison material, namely throughput, delay, packet loss, jitter, and using the TIPHON standard as a standard for assessing the quality of a network. However, the index values obtained in this experiment are dissected and compared more deeply. In that case, both methods have their respective advantages, namely that the HTB method tends to be superior if using TrueConf video conferencing or better than the PCQ method, where the difference can be seen in terms of throughput with an HTB index value of 3 and a PCQ index value of 2. while PCQ is superior to HTB when using Zoom video conferencing or better than HTB. The difference is also the same as before, only on the throughput side, with a PCQ index value of 3 and an HTB index of 2. With bandwidth management on both video conferences, there is a significant drop on the throughput side with an index value of 1, which is good. Although the difference in index values is not too large between the two methods, it needs to be more specific in practice to affect the performance of a network. Apart from the comparison of QoS, which way is best to use is also optional, depending on the user, because from another point of view, the advantages of a simpler implementation method or more complete features also affect the choice of the process you want to use.

5. Conflicts of Interest

The authors declare no conflicts of interest.

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