ANALYSIS QOS (QUALITY OF SERVICE) MEASUREMENT OF DELAY, JITTER, PACKET LOSS, THROUGHPUT, BANDWIDTH UTILITY AND RESOURCE OF USING ONLINE VIDEO CONFERENCING SOFTWARE

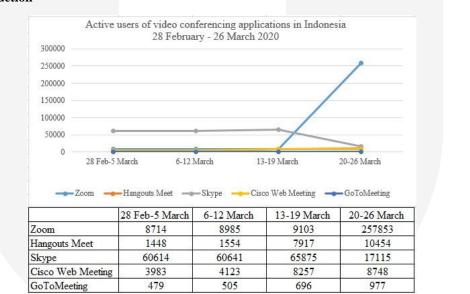
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Abstract

Video conferencing has become very popular due to the pandemic situation. In addition, working from home becomes the new normal for many employees. Thus, the use of video conferencing is higher and more intensive. There are plenty of video conferencing platforms that many people are using, such as Google Meet, Zoom, Big Blue Button, GoToMeeting, Cisco Webex, and Join.me. In order to overcome a large number of online video conferencing users from big companies to students who want to study, with the comparison of traffic bandwidth, security, and all resources in the video conference application. This journal means creating an efficient effort to reduce how to using video conferences based on QoS Parameter, utility bandwidth, and resource for the online video conference software. The test scenario lasts 5 minutes per session with five conditions measured on the traffic shaper, namely Normal conditions, 100Kbps limit, 50Kbps limit, 20% drop, and 50% drop. The measurement results show that Cisco Webex is more friendly to computers with low specifications, and Google Chrome, according to the TIPHON Standard, gets the highest index than other online video meeting software tested.

Keywords: Video Conference, Internet, Connection, Bandwidth



1. Introduction

Fig. 1 Active users of video conferencing applications in Indonesia 28 February - 26 March 2020.

The new type of Coronavirus disease pandemic 2019 or COVID-19 makes dif- ferent life in every world. Indonesia is the 4th largest population in the world, with a population of around 267 million. From such a large population, The total number of Indonesian workers as of August 2019 is 126.51 million people with a compar- ison of 70.49 million informal workers and 56.02 million formal workers. Due to a new policy that forces several companies to choose the concept of work from home (WFH) for their employees. Work from home is a term working remotely, or rather doing all office work from home. This work pattern, such as remote working, had occurred before COVID-19 existed and be in effect to employees with flexible working hours in Fig. 1. The increasing number of death caused by the COVID-19 from the World Health Organization (WHO) data increases public awareness of activities outside the home. In the early beginning of work from home trend or the isolation because of the Covid-19 outbreak, the world has been moving fast in using video conferencing [1]. There are Google Meet, Zoom, Big Blue Button, GoToMeeting, Cisco Webex and Join.me as examples. However, every client or user should consider the security, bandwidth, and the resource for using the online video conferencing apps [4].

Video conferencing helps many companies to find a solution in communicating with their workers, mostly that does not work from the office. However, some people are not quite sure about the safety of this video conferencing. Internet hacks and viruses can spread by dropping some threats in a chat room. Thus, as a user, it is essential to learn about how to video conferencing safely. It is good to have a chat room that requires a password to enter the chat room. It is a filter that will prevent non-member members from joining the chatroom.

2. Basic Concept

2.1 Wireshark

Wireshark is a network protocol analyzer that can track what is going on in the network or make sure the network is working correctly [3]. Wireshark works by scanning all packets passing through the selected interface such as networks-ethernets, Wi-Fi, monitor mode and Bluetooth.

2.2 Traffic Shaping

Regulate the transmission of incoming packets or delay the flow of certain types of network packets to ensure network performance for higher priority applications. Net Balancer is a monitoring application for all conditions on the network and limits downloading and uploading an application and network control that can be easily monitoring.

3. System Design

This journal aims to collect traffic or data such as jitter, delay, throughput, and bandwidth from each online video conference to analyze which software is better to use in bandwidth, security, and resource utility. In this thesis, each explanation will focus on every online video conference data output that has been processed on Wireshark [2]. In general, Wireshark is used in analyzing network performance by capturing packet transmission, as shown in Fig. 2. In this thesis, Wireshark captures every traffic to and from the user to the web or online video conference software. The traffic data obtained shows how much bandwidth, throughput, delay, and jitter each software is.

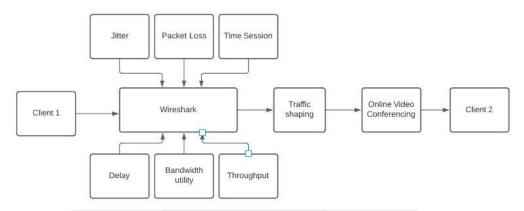


Fig. 2 Block Diagram System.

3.1 Block Diagram System

The system starts when the client is connected to the internet using a computer or laptop and accesses on- line video conference software such as Google Meet, Zoom, Big Blue Button, Go-ToMeeting, Cisco Webex and join.me using and interacting as usual and by using a modem connection with download speed and upload a minimum of 3Mbps, after connecting between client one and client two the network traffic is recorded on the WireShark and where traffic is selected from and to the online video conference.

3.2 Measurement

3.2.1 TIPHON

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) is a standard for evaluating QoS parameters issued by the ETSI (Euro- pean Telecommunications Standards Institute)[6] standards body which includes:

- 1. General information end-to-end quality and affected by various components in TIPHON system
- 2. Four classes in TIHPON Quality of Service used to classify TIPHON service
- 3. Description and relationship of performance on terminals and TIPHON Qual- ity of Service classes
- 4. The description of how the TIPHON system, terminal, and network perfor- mance can be measured.

3.2.2 Throughput

Wireshark is used to measure the throughput. Throughput is the actual band- width which is measured by a specific unit of time that is used to transfer data with a specific size. In Fig. 3 Througput is the same definition as Average bits/s on Wireshark.

In this case, the traffic that is counted is only through the Online video confer- ence, so the calculated throughput must go through filtering traffic first and then calculated.

Statistics			
Measurement	Captured	Displayed	Marked
Packets	96843	96843 (100.0%)	
Time span, s	299.902	299.902	1000
Average pps	322.9	322.9	10000
Average packet size, B	785	785	3 <u>88</u>
Bytes	76015123	76015123 (100.0%)	0
Average bytes/s	253k	253k	_
Average bits/s	2027k	2027k	

Fig. 3 Wireshark statistic of Zoom online video meeting.

3.2.3 Delay

Delay is when it takes for a packet to be sent from a client to the destination client. A long queue causes the delay in transmitting a packet, or it can take another route further from the original route to avoid congestion in routing. Delay can be found by calculating between the first packet and the last packet. For the formula

can be shown:

Average delay =
$$\frac{T.stmp2 - T.stmp1}{Number o f Packets}$$
(1)

3.2.4 Packet Loss

Packet loss is the percentage of packets lost during data transmission or a pa- rameter that indicates the number of packets lost [5]. Packet loss occurs due to several factors including decreased signal in the network media, overload, errors that occur on the physical media, and failure can also occur at the receiving end. Determine packet loss in Wireshark using the filter tcp.analysis.lost segment shown in Fig. 3.5 or by directly opening the statistics section on the capture file properties in the interfaces dropped packets on Wireshark are the same as packet loss shown in Fig. 4.

nterface	Dropped packets	Capture filter	Link type	Packet size lim
Ithernet	0 (0.0%)	none	Ethernet	262144 bytes

Fig. 4 Packet Loss shown in Capture File Properties.

3.2.5 Jitter

Jitter is defined as delay variation caused by queue length in data processing and reassembly of data packets at the end of transmission due to previous failures. Jitter also means the difference between the first delay and the next delay; if the delay variation in packet transmission is too wide, it will affect the transmitted data's quality. The amount of jitter tolerance in the network is affected by the number of buffers, which can reduce jitter effects. For the formula can be shown:

$$Jitter = \frac{Total \ delay \ variat \ ion}{Total \ packages \ received - 1}$$
(2)

3.2.6 Bandwidth utility

In this case, the bandwidth used is between 3Mbps and 6Mbps at the time of data measurement; the bandwidth used is the average bandwidth obtained to calcu- late the bandwidth utilization. Bandwidth utility is the percentage amount obtained from the average throughput divided by the provider's bandwidth to obtain the band- width efficiency value consumed during packet transmission. For the formula can be shown:

	Average Throughput		
Bandwidth utility $=$	Provider Bandwidth	-100%	(3)

3.2.7 CPU and Memory utilization

Host and Client use different hardware and OS (operating system), so CPU and memory usage will be different. Memory usage depends on how much memory capacity is available. Client 1 (Host) using 8 GB of memory and Client 2 (Client) using 2 GB of memory will affect running the same program. Likewise, the CPU usage on the Host uses an Intel^R CoreTM i3-4150 processor, and the Client uses an Intel^R CoreTM i3-4150 processor so you will see the difference in the process of running an online video conference.

Checking CPU and Memory utilization can use the default Windows applica- tion, namely the Task manager. Task manager displays CPU Utilization, CPU Speed, Uptime, Memory, disk information and more. Host and Client use differ- ent hardware and default software so the comparison can be seen in Fig. 5.

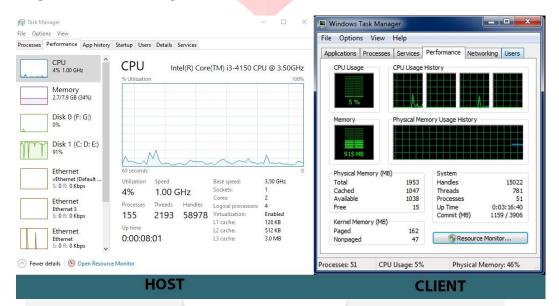


Fig. 5 CPU and Memory Monitoring on Host and Client.

4. Result and Analysis

4.1 Online Video Conference Software Compariso

4.1.1 Normal

Software	Total Packet (kB)	Throughput (kb/s)	Total Delay (s)	Average Delay (ms)
Zoom	696	18.66	298.54	105.08
Google Meet	61621	1641.37	300.34	2.39
GoToMeeting	4443	120.87	294.05	44.87
Cisco Webex	11309	302.32	299.25	13.53
BigBlueButton	26814	714.82	300.09	3.88
	Jitter (ms)	Packet Loss (%)	Bandwidth Utility (%)	Time Interval (s)
Zoom	0.06	0.07	0.31	298.54
Google Meet	0.00004	0.01	27.35	300.34
GoToMeeting	0.39	3.26	2.01	294.05
Cisco Webex	0.002	0.009	5.03	299.25
BigBlueButton	0.00008	0.01	11.91	300.09

Table 1 The measurement results on the Wireshark under normal conditions

This Table 1 shows that under normal conditions, all software is measuring on the Wireshark. The tiniest total packets are at Zoom Cloud meetings, and the most are Google meet. Zoom the least amount of total packets because it has its software, unlike Google Meet, which requires a browser to open it, and also needs to open a website and log in to a google account. In terms of the highest throughput is Google meet and the lowest is Zoom.

The highest average delay is a Zoom of 105.08ms, and the lowest is Google Meet with 2.39ms. The average time interval is 300 seconds or within 5 minutes. The highest packet loss was GoToMeeting, with 3.26% or equivalent to 214 packet loss from 4443042 packets received. The lowest was Google Meet with 0.019% and BigBlueButton with 0.010%; the lowest according to total packet loss was Big- BlueButton with only eight packet loss. The lowest bandwidth utility is at Zoom with 0.3% due to the small throughput compared to other software, and the biggest is Google Meet with 27.3%

4.1.2 100Kbps

Table 2 The measurement results on the Wireshark under 100Kbps conditions

Software	Total Packet (kB)	Throughput (kb/s)	Total Delay (s)	Average Delay (ms)
Zoom	696	18.66	298.54	105.08
Google Meet	61621	1641.37	300.34	2.39
GoToMeeting	4443	120.87	294.05	44.87
Cisco Webex	11309	302.32	299.25	13.53
BigBlueButton	26814	714.82	300.09	3.88
	Jitter (ms)	Packet Loss (%)	Bandwidth Utility (%)	Time Interval (s)
Zoom	Jitter (ms) 0.06	Packet Loss (%)0.07	Bandwidth Utility (%) 0.31	Time Interval (s) 298.54
Zoom Google Meet				
	0.06	0.07	0.31	298.54
Google Meet	0.06 0.00004	0.07 0.01	0.31 27.35	298.54 300.34
Google Meet GoToMeeting	0.06 0.00004 0.39	0.07 0.01 3.26	0.31 27.35 2.01	298.54 300.34 294.05

This Table 2 shows that in a limited condition of 100Kbps on the Netbal- ancer as measured by the Wireshark. The smallest total packet is still at Zoom, which is 622200bytes packet, and the most are Google Meet with a total packet of

65230988bytes; BigBlueButton has a total packet almost below Google Meet with a total of 52905346bytes. In terms of throughput, it is also the same as total packets, and the lowest is zoom, and the highest is google meet. The average time interval is

300 seconds or within 5 minutes.

The highest average delay at 100Kbps limit is 113.4ms at Zoom and the lowest in the BigBlueButton with an average delay of 3.07ms. In this table, the most Packet Loss is GoToMeeting with 0.8% or 55 total packet loss, and the lowest is 0.009% for BigBlueButton with only nine packet

loss. The most significant bandwidth utility on Google Meet, following the amount of throughput, is the utility bandwidth of 28.4%, and the lowest is Zoom at 0.2%.

4.1.3 50Kbps

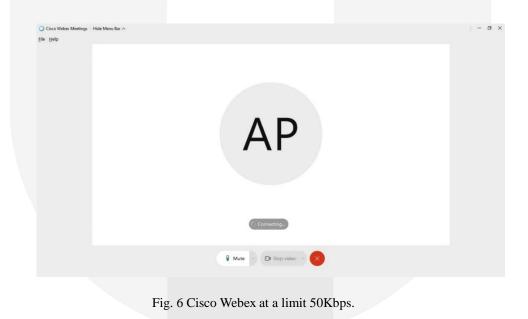
Table 3 The measurement results on the Wireshark under 50Kbps conditions

Software	Total Packet (kB)	Throughput (kb/s)	Total Delay (s)	Average Delay (ms)
Zoom	27679	738.12	299.99	2.95
Google Meet	34433	917.28	300.31	3.46
GoToMeeting	4241	114.26	296.93	10.35
Cisco Webex	2881	77.19	298.57	24.02
BigBlueButton	379	10.17	298.61	311.05
	Jitter (ms)	Packet Loss (%)	Bandwidth Utility (%)	Time Interval (s)
Zoom	0.009	0.01	12.30	299.9 <mark>9</mark>
Google Meet	0.00005	0.10	15.28	300.31
GoToMeeting	0.0001	0.19	1.90	296.9 <mark>3</mark>
Cisco Webex	0.004	0.07	1.28	298.57
BigBlueButton	0.17	0.41	0.16	298.61

This Table 3 shows that all limited applications are only 50Kbps to down- load and upload on NetBalancer and are measured on Wireshark. The smallest total packet changed from initially Zoom to regular and 100Kbps; the smallest to- tal packet is BigBlueButton with only 379939bytes following throughput of only

10178.73bit / s the highest Google meet and followed by Zoom with total packets of 34433841bytes and for Zoom is 27679641bytes. The average time interval is 300 seconds or within 5 minutes.

BigBlueButton's most significant average delay with 311.05ms, and the small- est is Zoom, with only 2.95ms. The minor packet loss is still at Zoom with 0.015% or 16packets, and the largest is 0.4% at BigBlueButton. The most significant band- width utility is still on Google Meet with 15%, and the lowest is BigBlueButton with 0.16% only. Cisco Webex cannot connect and will appear continuously recon- necting as shown on Fig. 6.



4.1.4 Drop 20%

Software	Total Packet (kB)	Throughput (kb/s)	Total Delay (s)	Average Delay (ms)
Zoom	41450	1106.33	299.72	2.70
Google Meet	64949	1731.71	300.04	2.92
GoToMeeting	187	5.35	280.17	702.18
Cisco Webex	13413	357.35	300.28	9.36
BigBlueButton	11355	302.91	299.54	5.78
	Jitter (ms)	Packet Loss (%)	Bandwidth Utility (%)	Time Interval (s)
Zoom	0.008	0.04	18.43	299.72
Google Meet	0.00001	0.13	28.86	<u>300.04</u>
GoToMeeting	0.58	1.00	0.08	280.17
Cisco Webex	0.002	0.13	5.95	<u>300.28</u>
BigBlueButton	0.00004	0.10	5.04	299.90

Table 4 The measurement results on the Wireshark under 20% conditions

20% drop that has been set in the NetBalancer software and measured using the Wireshark. Measurement results into several parameters shown on this Table 4, the highest total packet on average is still on Google Meet with a total packet of

64949948bytes. The lowest is for GoToMeeting with 187553bytes following the throughput directly proportional to the total packet in each software.

The most significant average delay in GoToMeeting is 702.1ms, and the lowest is at Zoom and Google Meet, which are 2.7ms and 2.9ms, respectively. The most packet loss is on GoToMeeting with 1,002% with a total packet loss of 4 packets, and the lowest is at zoom with only 0.04%. Time interval the average measurement is 300 seconds or 5 minutes.

GoToMeeting at a 20% drop can open the website, but after log on to a meeting, it just gets stuck on the loading screen to start the meeting as shown in Fig. 7. On drop 20%, the Cisco Webex meeting cannot connect to join a meeting as shown in Fig. 8.



Fig. 7 GoToMeeting at a 20% drop.

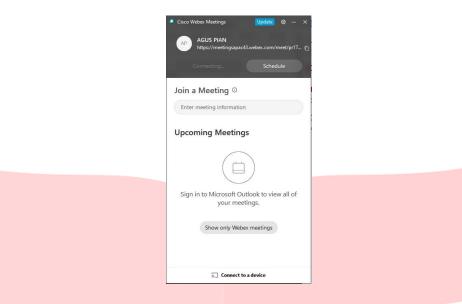


Fig. 8 GoToMeeting at a 20% drop.

4.1.5 Drop 50%

Table 5 The measurement results on the Wireshark under 50% conditions

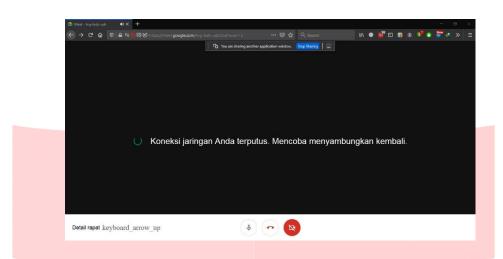
Software	Total Packet (kB)	Throughput (kb/s)	Total Delay (s)	Average Delay (ms)
Zoom	345	9.21	299.59	155.07
Google Meet	26953	718.74	300	5.24
GoToMeeting	266	7.26	293.50	448.09
Cisco Webex	1857	50.57	293.85	74.24
BigBlueButton	40	3.23	101.44	1193.50
	Jitter (ms)	Packet Loss (%)	Bandwidth Utility (%)	Time Interval (s)
Zoom	0.01	0.20	0.15	299.59
Google Meet	0	0.06	11.97	300
GoToMeeting	0.009	1.06	0.12	293.50
Cisco Webex	0.005	0.42	0.84	293.85
BigBlueButton	2.74	3.52	0.05	101.44

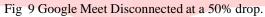
50% drop in setting on NetBalancer and measured in Wireshark. The most sig- nificant total packets are still on Google Meet with 26953670bytes, and the lowest is at BigBlueButton 40982bytes, for throughput is directly proportional to the total packets.

The most average delay is BigBlueButton with an average of 1193.505471ms, and the least is google meet with 5.2ms. Bandwidth utility is also directly pro- portional to the number of packets and throughput with the highest and lowest on Google meet and BigBlueButton. The highest packet loss is also at BigBlueButton with a massive 3.5%, inversely proportional to Google Meet, which is only 0.06% as shown as Table 5.

The average time interval for all measurements is 300 seconds, but for measure- ments with a 50% drop condition, the BigBlueButton time interval only reaches

101.4 seconds. In this 50% drop condition, all software cannot be connected and run correctly. Google meet to enter the google web, log into a google account, cre- ate a room, and enter an online meeting room. However, after entering, a network connection is always disconnected, like in this Fig. 9. Zoom Unable to connect with error code: 5003 as shown in this Fig. 10. At BigBlueButton cannot load the website to enter the meeting room seen in this Fig. 11. GoToMeeting can enter the website, but to enter the meeting room, it will load continuously, as shown in this Fig. 12. Cisco Webex cannot join to conduct a meeting. Reconnecting will appear continuously, as in this Fig. 13.





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Fig	. 10 Zoom Error	Code and	Un	able to	o Conne	ect at a 50%	drop.
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Fig. 11 BigBlueButton Unable to Load at a 50% drop.

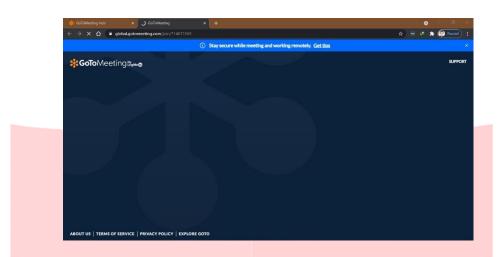


Fig. 12 GoToMeeting stuck when loading the website at a 50% drop.

Q Cisco Webex Meetings Hide Menu Bar ∧ Elle Help		
	AP	
	() Connecting	
	🖉 Mute 🖂 🕞 Stop video 🕞 🗙	
	With a Dissipance	

Fig. 13 Cisco Webex cannot join to conduct a meeting at a 50% drop.

4.2 CPU AND Memory Utilization

Measurement results on CPU and memory usage to find out which online video conference software is better in memory usage and also CPU resources in the device used in the system design. CPU and memory measurements use a task manager application with Windows 7 (Client) and Windows 10 (Host) OS when the computer is idle and opening the online video conference software.

4.2.1 GoToMeeting

Table 6 GoToMeeting Cpu and Me	emory Utilization
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Demonstern		Client			Host	
Parameter	Idle	Open	Difference	Idle	Open	Difference
Cpu Usage	0%	18%	18%	3%	51%	48%
Memory	703MB	1.10GB	397MB	3.9GB	4.7GB	0.8GB

The table 6 shows CPU usage in Client of 18% and memory usage of 397MB from the idle condition at 703MB to 1.10GB when the GoToMeeting software is open.

The table on Host tends to be greater due to using 64bit software with CPU Usage reaching 48% from the initial condition 3% to 51% when opening the Go- ToMeeting software. The memory usage reaches 0.8GB from the idle condition at 3.9GB. When opening the software, it reaches 4.7GB, and this is because the GoToMeetings Software requires browsers to open the GoToMeeting, which makes high memory resource usage.

4.2.2 Google Meet

Table 7 Google Meet Cpu and Memory Utilization

Parameter		Client		Host			
	Idle	Open	Difference	Idle	Open	Difference	
Cpu Usage	0%	28%	28%	2%	30%	28%	
Memory	794MB 1.40GB		606MB	3.7GB 4.8GB		1.1GB	

The data in the table 7 shows that Google Meet on a Client consumes up to 28% of memory from idle at 0%, and when opening Google Meet, it increases to 28%. Open google meets entirely takes up much memory because it uses the Browser to open google meet by consuming 606MB from idle at 794MB to 1.40GB to open the Firefox browser with one open tab Google Meet website.

The use of Google Meet on Host consumes up to 1.1GB of memory from idle at

3.7GB. When opening the Browser and opening the tab for the google meet website, the memory reaches 4.8GB from the available 8GB memory. Google Meet also consumes CPU Usage the same as Client, which is 28% of the idle state at 2% and reaches 30% when opening Google meet.

4.2.3 Zoom Online Video Meetings

Table 8 Zoom Cpu and Memory Utilization

Parameter		Client		Host				
	Idle	Open	Difference	Idle	Open	Difference		
Cpu Usage	1%	29%	28%	8%	9%	1%		
Memory	Memory 532MB		704MB 172MB		4.3GB	0.1GB		

From the data in the table 8, the use of the Zoom software on a Client takes up 172 MB of memory, from idle at 532 MB to 704 MB when opening the Zoom application and consuming 28% CPU Usage, i.e., from idle at 1% and when opening CPU usage software rose to 29%.

The use of Zoom software on Host consumes minimal CPU Usage, which is only 1% of it is idle at 8%, and when opening the software, it only increases to 9%. Memory usage in the Zoom software is also relatively small, namely 0.1GB, the same as memory usage in Client; this makes Zoom the lightest testing software on hardware conditions such as Host.

4.2.4 Big Blue Button

Table 9 BigBlueButton Cpu and Memory Utilization

Parameter		Client		Host			
	Idle	Open	Difference	Idle	Open	Difference	
Cpu Usage	0%	24%	24%	2%	17%	15%	
Memory	826MB	952MB	126MB	3.6GB	4.1GB	0.5GB	

The table 9 shows that BigBlueButton's CPU usage reaches 24%, and mem- ory usage is only 126MB on the Client with idle condition 826MB to 952MB when the software is open.

In the table on Host, CPU usage is only 15%, from idle 2% becomes 17% when BigBlueButton is open. This software requires a browser available on the Win- dows version to run it; memory usage only requires 0.5GB from 8GB available from 3.6GB idle condition to 4.1GB when opening the Browser and opening the Bigbluebutton Web.

4.2.5 Cisco Webex

Table 10 Cisco Webex Cpu and Memory Utilization

	Parameter		Client		Host			
		Idle	Open	Difference	Idle	Open	Difference	
	Cpu Usage	0%	7%	7%	3%	29%	26%	
	Memory	727MB	844MB	117MB	3.9GB	4.3GB	0.4GB	

The table 10 shows the lowest CPU Usage on Cisco Webex on Client, which is only 7%, and also for memory usage, which is 117MB from idle at 727MB and when opening software at 844MB only makes Cisco Webex the lightest to use with low specifications such as hardware. Client The table shows that the CPU Usage used is 26%, with idle conditions at 3%. When opening the software to 29% and memory usage, it takes up 0.4GB from idle at 3.9GB to 4.3GB when opening the software, relatively light of some software under test.

4.3 Telecommunications and Internet Protocol Harmonization Over Networks **4.3.1** Zoom Online Video Meetings

Table 11 The Measurement Results of the Zoom Cloud Meeting Parameters with the TIPHON Standard

Parameter	Normal	TIPHON Standard	100Kbps	TIPHON Standard 2	50Kbps	TIPHON Standard 3	Drop 20%	TIPHON Standard 4	Drop 50%	TIPHON Standard 5
Throughput (bps)	18663.3427		16644.61607		738128.0018		1106335.87		9213.825078	
	0.3%	Poor	0.2%	Poor	12.3%	Poor	18.4%	Poor	0.1%	Poor
Average Delay (ms)	105.0835787	Perfect	113.4060129	Perfect	2.954716503	Perfect	2.702426468	Perfect	155.0712283	Good
Jitter(ms)	0.069958817	Good	0.092833	Good	0.009845	Good	0.008931	Good	0.015147	Good
Packet Loss (%)	0.070397747	Perfect	0.037921881	Perfect	0.015758579	Perfect	0.041474696	Perfect	0.207039337	Perfect

The Table 11 shows that meeting using Zoom show that all measurements on the Packet Loss parameter are all indexed four or perfect based on the TIPHON standard, as well as the Delay parameter with index four, or Perfect except for measurements with Zoom conditions with 50% Drop getting index three or Good according to the TIPHON Standard. The jitter parameter in all measurements under any circumstances gets an index of 3 or Good. The throughput of each measurement in the Zoom software is all indexed one or poor based on the TIPHON standard because it is below 25%.

The figure shows that the Index of Normal Condition, 100Kbps, 50Kbps, and

20% Drop shows an average index of 3 which means Good except for the 50% drop, which shows the average index value is 2.75.

4.3.2 GoToMeeting

Table 12 The Measurement Results of the GoToMeeting Parameters with the TIPHON Standard

Parameter	Normal	TIPHON Standard	100Kbps	TIPHON Standard 2	50Kbps	TIPHON Standard 3	Drop 20%	TIPHON Standard 4	Drop 50%	TIPHON Standard 5
Throughput (bps)	120878.4725		70375.85563		114264.571		5355.333336		7261.651548	
	2%	Poor	1.1%	Poor	1.9%	Poor	0.08%	Poor	0.12%	Poor
Average Delay (ms)	44.87260415	Perfect	44.89457024	Perfect	10.35517817	Perfect	702.1899323	Poor	448.0918702	Medium
Jitter(ms)	0.39058218	Good	0.03016122	Good	0.00015153	Good	0.58062657	Good	0.00959542	Good
Packet Loss (%)	3.265679841	Good	0.832576446	Perfect	0.198779425	Perfect	1.002506266	Perfect	1.06870229	Perfect

The Table 12 shows that when using GoToMeeting, all throughput shows in- dex 1, which means Poor in every situation. Contrary to Jitter and packet loss, respectively index 3 and 4, which means Good and Perfect, parameter delay index

4 occurs in Normal, 100Kbps and 50Kbps. For Drop 20% and 50% only index 1 and 2 only. The figure explains that the average conditions during a GoToMeeting meeting at the time of measurement showed Good and under, where the highest av- erage index is only three at 100% Limit conditions. At 50% for normal conditions, it only reaches an index of 2.75 and Drops 20% and 50%.

4.3.3 Google Meet

Table 13 The Measurement Results of the Google Meet Parameters with the TIPHON Standard

13

Parameter	Normal	TIPHON Standard	100Kbps	TIPHON Standard 2	50Kbps	TIPHON Standard 3	Drop 20%	TIPHON Standard 4	Drop 50%	TIPHON Standard 5
Throughput (bps)	1641370.475		1737270.217		917280.501		1731715.657		718742.9183	
	27.8%	Medium	28.9%	Medium	15.2%	Poor	28.8	Medium	11.9	Poor
Average Delay (ms)	2.391375857	Perfect	3.163800166	Perfect	3.469212973	Perfect	2.922460495	Perfect	5.240515337	Perfect
Jitter(ms)	0.00004282	Good	0.00003337	Good	0.00005455	Good	0.00001787	Good	0.000	Perfect
Packet Loss (%)	0.019109345	Perfect	0.053715875	Perfect	0.102812915	Perfect	0.133437226	Perfect	0.064631079	Perfect

The Table 13 shows that meeting using Google Meet shows that all mea- surements on the Delay and Packet Loss parameter are four, which means that all are perfect according to TIPHON Standard. Throughput in Normal conditions,

100Kbps and 20% drop index two, which means Medium, and at 50Kbps and Drop

50% indicates index one, which means Poor. The figure shows that the average use of Google Meet in the conditions taken when the measurement shows that every- thing is good. The average obtained is 3.25 in each condition except the average at

50Kbps, which gets index three and is still in good condition according to TIPHON Standard.

4.3.4 Big Blue Button

 Table 14 The Measurement Results of the BigBlueButton Parameters with the TIPHON

 Standard

Parameter	Normal	TIPHON Standard	100Kbps	TIPHON Standard 2	50Kbps	TIPHON Standard 3	Drop 20%	TIPHON Standard 4	Drop 50%	TIPHON Standard 5
Throughput (bps)	714823.374		1409445.489		10178.73769		302915.5559		3231.765171	
	11.9%	Poor	23.4%	Poor	0.16%	Poor	5.04%	Poor	0.05%	Poor
Average Delay (ms)	3.888282755	Perfect	3.077154447	Perfect	311.0560885	Medium	5.789147346	Perfect	1193.505471	Poor
Jitter(ms)	0.00008695	Good	0.00006404	Good	0.17687812	Good	0.00000462	Good	2.74450588	Good
Packet Loss (%)	0.01036538	Perfect	0.00922254	Perfect	0.416666667	Perfect	0.10436396	Perfect	3.529411765	Good

The Table 14 shows that when using Bigbluebutton for meetings in all condi- tions, get an average Throughput Poor or index 1, for Jitter and Packet Loss get the same index, namely 3 and 4 in each condition Good and Perfect. The delay param- eter shows the difference in the 50Kbps limit condition and the 50% Drop, which only gets index 2 and 1 respectively, which means Medium and Poor. Others get four each for Normal, 100Kbps conditions, and at 20% Drop. The figure shows that meetings using Bigbluebutton on average get index 3, which means Good accord- ing to the TIPHON Standard for all measurement in Normal conditions, 100Kbps Limit, 20% Drop and 50% Drop, but for Limit 50Kbps conditions only get 2.5 index due to delay in this condition is index 2.

4.3.5 Cisco Webex

Table 15 The Measurement Results of the Cisco Webex Parameters with the TIPHON Standard

Parameter	Normal	TIPHON Standard	100Kbps	TIPHON Standard 2	50Kbps	TIPHON Standard 3	Drop 20%	TIPHON Standard 4	Drop 50%	TIPHON Standard 5
Throughput (bps)	302326.4413		72439.03413		77195.65787		357358.8815		50576.16907	
	5.03%	Poor	1.2%	Poor	1.28%	Poor	5.9%	Poor	0.84%	Poor
Average Delay (ms)	13.5390484	Perfect	23.5834529	Perfect	24.0223083	Perfect	9.3605525	Perfect	74.2431241	Perfect
Jitter(ms)	0.00235787	Good	0.00029185	Good	0.00448170	Good	0.00242347	Good	0.00567863	Good
Packet Loss (%)	0.009048545	Perfect	0.157629256	Perfect	0.072411296	Perfect	0.130922693	Perfect	0.429509853	Perfect

The Table 15 shows that all parameters show the same index in every condition in a meeting using Cisco Webex. The throughput parameter is stagnant at one, which means Poor based on TIPHON Standard. The Delay parameter shows the Perfect index: the value 4, Jitter index three, and Packet Loss indexed 4 in all circumstances from Normal, limit 100Kbps, Limit 50Kbps, Drop 20%, and Drop

50%. The figure shows that the average in each measurement under any conditions all get the same index, which is three, which means Good according to the TIPHON Standard.

5. Conclusion

This measurement aims to compare online video conference software based on the influence on specific parameters and conditions to test video conference software in an online meeting activity with a duration of 5 minutes. The highest index obtained by video conferencing software based on the TIPHON standard is Google with an average index of 3.2, and the lowest index is on Bigbluebutton. Computer-friendly online video conferencing software is Zoom because it is lighter on CPU and Memory Usage than other software. As for future studies, it is Essential to analyzing more conditions and features than net balancer software as a medium for traffic shaping.

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