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Feasibility Study of Universitas Advent Indonesia Network for Hybrid Distance Learning

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ABSTRACT

The rapid development of the internet has spurred many fields to adopt remote meetings. Indonesia Adventist University (UNAI) has the capability to implement hybrid learning, particularly in distance learning. In this research, testing will be conducted at several student testbed locations across UNAI, focusing on various QoS (Quality of Service) parameters, including bandwidth, delay, jitter, and UDP packet loss rate. The results show significant variations in QoS across different testbeds, presenting specific challenges for future experimental plans.

Keywords: Network Readiness, QOS, Hybrid Learning, Distance Learning

INTRODUCTION

The rapid development of technology has allowed many fields to operate more freely and reach an increasing number of users via the Internet. One of the triggers for the rapid business transition in adopting technology was the 2020 pandemic. This swift transition also occurred at Universitas Advent Indonesia, where a strategic decision had to be made regarding the adoption of technology for distance learning (Limbong, 2022). Some approaches taken included using Zoom for meetings and Moodle as the Learning Management System. In several previous studies, it has been proven that Zoom has been instrumental in facilitating teaching and learning activities (Boy et al., 2022; Iryani et al., 2020).

Some challenges encountered during distance learning include interrupted or hindered media transmission due to varying quality of service in different regions. In previous research conducted by (Sihotang & Bandung, 2016), Background traffic has a significant impact on the quality of multimedia services, especially for online classes that use the UDP protocol with a higher bitrate compared to text-based services that use the TCP protocol. This prompted researchers, at the end of the pandemic, to conduct a study on the readiness of Universitas Advent Indonesia to handle hybrid distance learning (offline classes with some attendees joining from another city via online meetings). This research involved several testbeds

distributed from the western to eastern parts of Indonesia, and it is expected to provide a comprehensive overview of the network conditions at those testbed locations.

METHODS

In conducting the research, the following are the research stages used:

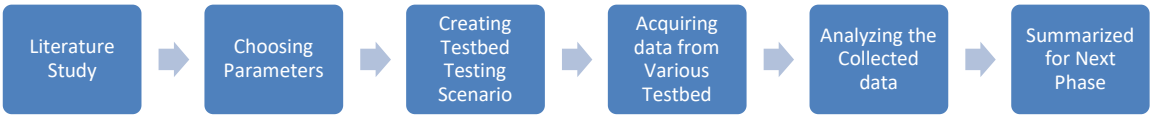

Figure 1 Research Methods

The initial stage begins with studying several pieces of literature from previous research, which are summarized into the background that motivates the study. This is then distributed across the document in the form of definitions or explanations for each research instrument.

QOS Parameters And Testbed Scenario

In collecting data from 10 testbeds spread across various locations in Indonesia, including Bandung, Jakarta, Bekasi, East Java, Kalimantan, Pekanbaru, North Sumatra, and Kupang. The data was collected remotely using AnyDesk to connect to volunteer computers at those locations. Several Quality of Service (QoS) parameters were used in this research. These QoS parameters include Bandwidth, Delay, Jitter, and UDP Packet Loss Rate (PLR) (Sigeti et al., 2013).

Each QoS test followed the scenarios below:

1. Data was collected on four different weekdays (08:00 AM – 05:00 PM) during periods when students were attending online classes via Zoom.
2. Additional data was collected during non-peak periods over four time periods on the same days during the weekend (or other days when students had no classes).

The measurement of QoS was conducted using Iperf/Jperf and PING (Packet Internet Gopher). Below is the explanation of the scenario for measuring each QoS parameter.

1. Bandwidth

   Bandwidth refers to the maximum data transfer rate of a network or internet connection, indicating how much data can be sent or received per second. It is typically measured in bits per second (bps), with higher bandwidth allowing more data to flow through the connection. While higher bandwidth can improve speed and performance, it is just one factor in overall network quality, alongside latency and reliability. Bandwidth capacity was measured using TCP traffic. TCP datagrams were sent using the Iperf/Jperf tool. To run this scenario, one host was needed as a client connected to the DSL internet

network and one server located at UNAI. Each measurement period was conducted for 10 seconds. The Iperf server could be accessed starting at 08:00 AM and was turned off at 05:00 PM. Volunteers at each testbed could perform the tests according to the scenario after confirming that they were ready to conduct the testing.

2. Delay

   Delay refers to the time it takes for data to travel from the source to the destination in a network. It is typically measured in milliseconds (ms) and can be affected by factors like distance, network congestion, and the number of intermediate devices (such as routers) along the path. High delay, often called latency, can cause noticeable lags in communication, especially in real-time applications like video calls or online gaming. E2E Delay is measured using ICMP request and reply to packets (Ping) (ITU, 1994). To obtain more accurate data, the measurement is performed several times with different packet sizes. The packet sizes sent are 1350 bytes (average Max MTU of the Testbed network) and 40 bytes (as referenced in the Dischinger paper). Each time the measurement is performed, 100 packets are sent. Like bandwidth, volunteers in each testbed can conduct tests according to the scenario after confirming their participation in the testing.

3. Jitter & UDP Packet Loss Rate

   Jitter refers to the variation in packet arrival times in a network, meaning that data packets may arrive at irregular intervals rather than a steady, predictable rate. And UDP Packet Loss Rate is the percentage of data packets lost during transmission over a network using the User Datagram Protocol (UDP) (Bekele et al., 2024). A high value of Jitter and/or UDP Packet Loss Rate can cause disruptions in real time communications. Jitter and UDP packet loss rate measurements are conducted using UDP traffic. UDP datagrams are sent using Iperf tools. To run this scenario, 2 hosts are required: one acts as the server (located at UNAI) and the other as the client (located at each testbed). Each measurement is performed for 10 seconds, and each measurement period lasts for 10 seconds. The Iperf server is accessible from 08:00 to 17:00. Volunteers in each testbed can perform the tests according to the scenario after confirming their participation in the testing.

Each set of data collected from the sessions is presented in table form to directly assess the quality of service for each testbed. The presentation of the measurement results from each testbed will be discussed in the next section.

RESULTS AND DISCUSSION

The results from the field tests or testbed, conducted at the end of 2021 with the assistance of several student volunteers who were willing to be remotely managed, and

following various specific scenarios and stages for each tested parameter, are presented in the following section.

Bandwidth

The bandwidth testing was done in various time during each Weekday or Weekend scenario as mentioned in previous section. Bandwidth is a primary parameter commonly understood by the public as a measure of network quality. This has led to the paradigm among the public that the higher the bandwidth, the better the ISP network. Bandwidth testing is conducted over several periods, with each testing period consisting of 4 sequences. The Table 1 below refer to the Bandwidth Testing of each testbed in Weekdays (the test performed while student having their online meeting using zoom).

Table 1 Bandwidth – On Class (Mbps)

Bandwidth - On Class (Busy)					
Testbed	**Seq. 1**	**Seq. 2**	**Seq. 3**	**Sec. 4**	**Average**
1	3,55	3,1	3,37	3,77	**3,45**
2	1,27	0,973	1,26	1,21	**1,18**
3	4,5	3,66	0,419	3,44	**3,00**
4	30,5	28,8	33,5	33,4	**31,55**
5	3,59	1,9	3,71	3,47	**3,17**
6	25,7	26,5	26	25,2	**25,85**
7	4,3	3,73	3,92	2,23	**3,55**
8	7,26	3,09	3,38	5,3	**4,76**
9	2,41	2,45	0,349	3,23	**2,11**
10	2,28	0,921	0,587	0,944	**1,18**

And now, to also measure the available bandwidth on weekends or during off-class periods, the same test is conducted to get an overview of the bandwidth availability during free time or when no Zoom classes are being held. Table 2 below provides an overview of the available bandwidth during off-class times or weekends.

Table 2 Bandwidth – Off Class (Mbps)

Bandwidth - Off Class (Free)					
Testbed	**Seq. 1**	**Seq. 2**	**Seq. 3**	**Sec. 4**	**Average**
1	3,67	3,98	3,35	4,09	**3,77**
2	2,83	1,62	2,25	0,639	**1,83**
3	1,47	1,36	1,68	1,47	**1,50**
4	33,5	33,4	33,4	33,4	**33,43**
5	3,74	3,34	3,28	2,75	**3,28**
6	25,3	25,1	26,5	26,4	**25,83**
7	4,29	4,3	4,3	4,27	**4,29**
8	7,65	11	7,74	8,53	**8,73**
9	3,1	3,75	2,48	3,39	**3,18**
10	0,807	2,31	2,31	2,35	**1,94**

When comparing the bandwidth availability during periods of Zoom meetings or when no classes are being held, there are several differences. This also does not account for background traffic that may occur during testing due to other users accessing the network at

the same time. The highest and most consistent bandwidth was found in Testbed 4 located in Bekasi, while the lowest bandwidth was recorded in Testbed 10 in Sidoarjo (followed by Testbed 2 located in North Sumatra). Figure 2 provides an infographic showing the differences in bandwidth between the two testing periods.

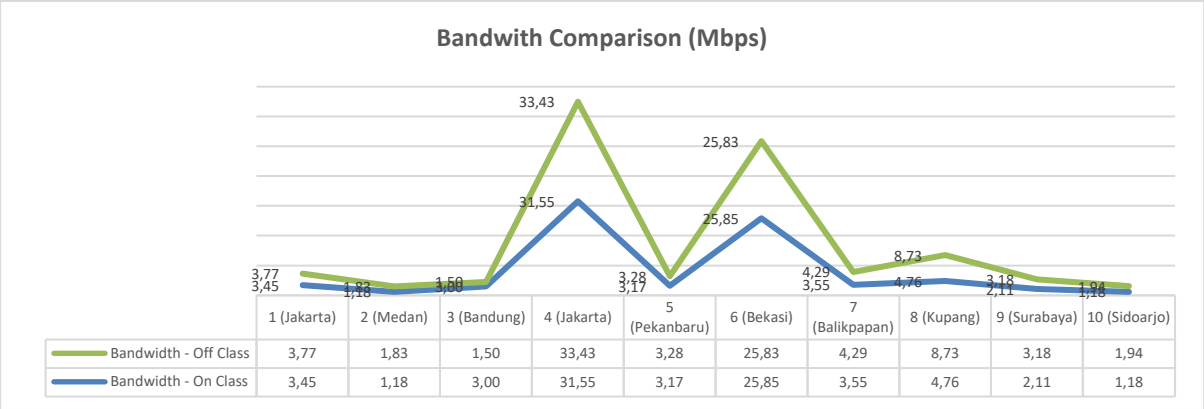

Figure 2 Bandwidth Comparison (Mbps)

Delay

Delay testing is conducted after the iperf command is executed in the bandwidth test, following each testing sequence in different periods. The packet size values used are 40 bytes (default) and 1350 bytes (Average Max MTU). Table 3 below shows the differences in delay between testing scenarios during Zoom classes (On Class) and when no classes are being held or during weekends (Off Class) for each TestBed (TB).

Table 3 Delay Comparisons (ms)

	TB-1	TB-2	TB-3	TB-4	TB-5	TB-6	TB-7	TB-8	TB-9	TB-10	Average
Delay (40) - On Class	9,25	118	130	6	49,75	8,75	64,5	151,25	149,75	444,75	**113,2**
Delay (1350) - On Class	10,5	140,5	163,25	7,25	53,25	9,75	66	152,75	192,75	381	**117,7**
Delay (40) - Off Class	7	153,75	118	6,75	47,75	12	39	96	115	71,25	**66,65**
Delay (1350) - Off Class	8	158,25	157,75	8	49,5	17	41,75	119,5	126,5	69,75	**75,6**

From the table above, it can be concluded that, in general, the level of delay increases when testing is conducted during class hours using Zoom meetings. Although testing outside class hours is not free from other traffic on the network that directly affects delay values. On average, the delay values in each testbed still fall within a good classification (below 150 ms) (International Telecommunication Union, 2001). However, it is noted that some testbeds have unacceptable delay values, especially when compounded by the load of meetings conducted via Zoom.

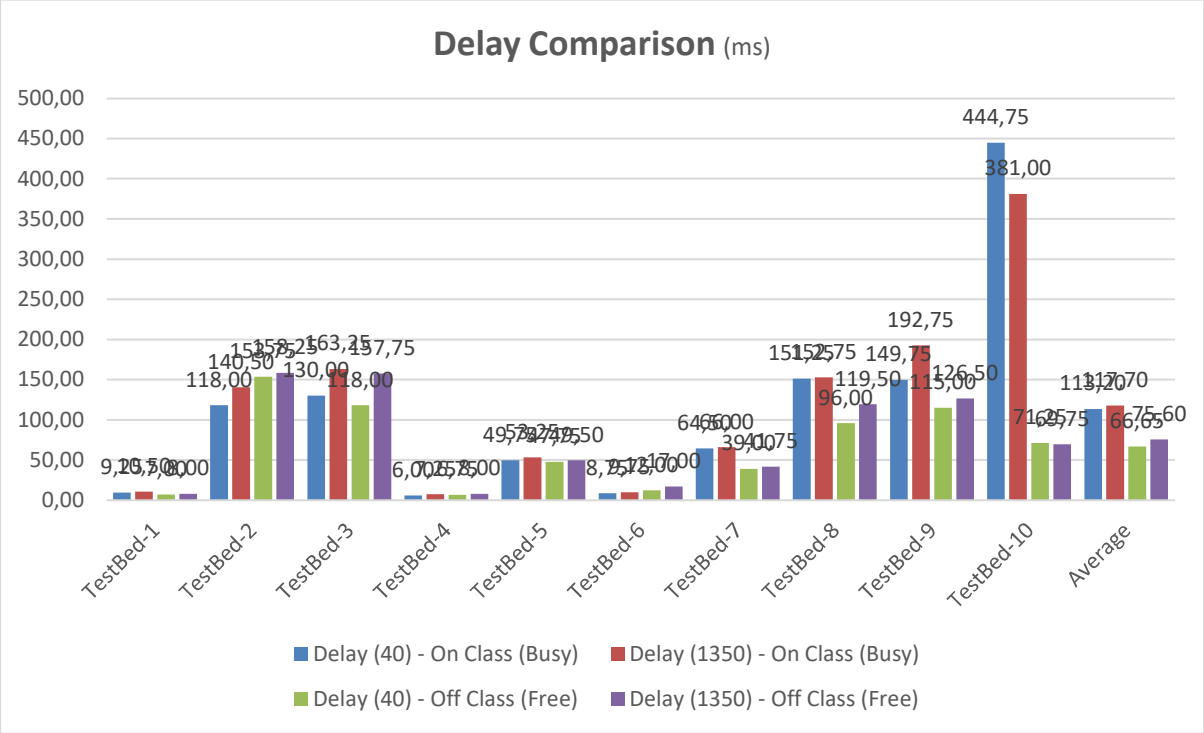

Figure 3 Delay Comparison (ms)

Jitter

Immediately after measuring the delay for both 40 and 1350 bytes, jitter measurements were conducted using the UDP protocol at 256K. The tests, performed during different periods—class time and off-class time—showed differences consistent with the previous two parameters tested. The testing was still conducted over 4 sequences in each testing period. Table 4 below summarizes the jitter testing for each scenario in each TestBed (TB).

Table 4 Jitter Comparison (ms)

	TB-1	TB-2	TB-3	TB-4	TB-5	TB-6	TB-7	TB-8	TB-9	TB-10	Average
Jitter – UDP - 256K – On Class	0,57	58,16	28,30	0,07	19,44	2,13	23,13	15,78	26,05	0,46	**17,41**
Jitter – UDP - 256K - Off Class	0,48	22,09	21,71	0,07	12,18	0,52	11,98	9,13	18,24	0,52	**9,69**

Consistent with the previous results, the condition of the network when burdened with online classes via Zoom increases the jitter values. However, on average, the jitter values are still considered acceptable (International Telecommunication Union, 2001). It is noted that Testbed 2 has a high jitter value that falls into the unacceptable category.

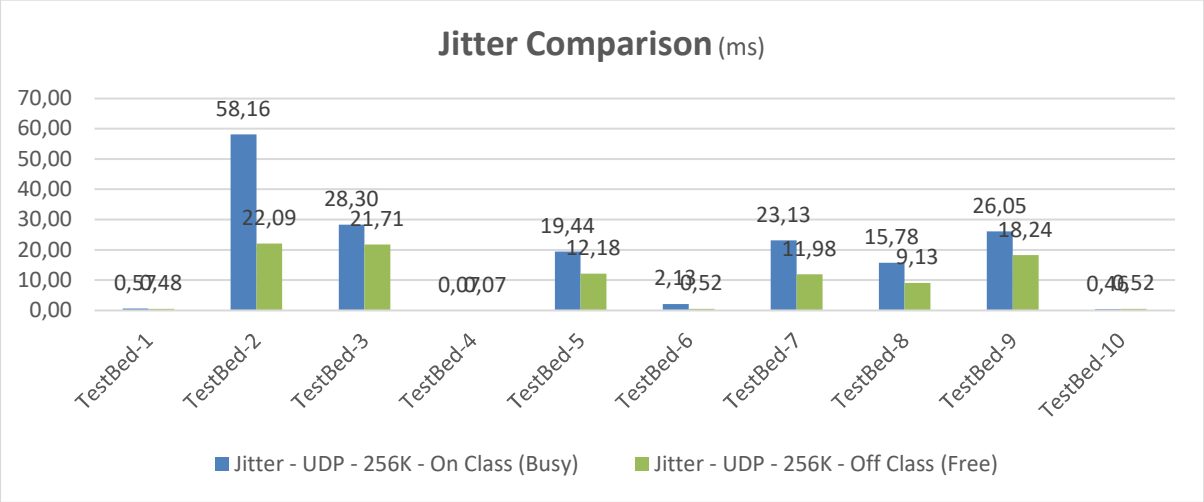

Figure 4 Jitter Comparison (ms)

UDP Packet Loss Rate

UDP Packet Loss Rate measures the number of datagrams received from each data transmission session. UDP is the protocol used for transmitting both static and dynamic images. Below is a comparison of the UDP Packet Loss Rate from each TestBed (TB) for the On Class and Off Class scenarios. One period of the scenario consists of 4 testing sequences.

Table 5 UDP Packet Loss Rate Comparison (%)

	TB-1	TB-2	TB-3	TB-4	TB-5	TB-6	TB-7	TB-8	TB-9	TB-10	Average
Packet Loss Rate - UDP - 256K - On Class	10%	49%	1%	32%	8%	0%	27%	33%	20%	57%	**24%**
Packet Loss Rate - UDP - 256K - Off Class	16%	14%	1%	17%	20%	0%	35%	0%	25%	55%	**18%**

From the test results above, the overall UDP packet loss rate is relatively high, especially in several testbeds with low bandwidth. In contrast, some testbeds with high bandwidth (e.g., TB-3, TB-6) experience very minimal datagram loss. Meanwhile, Testbed-10 demonstrates poor performance, with a UDP Packet Loss Rate exceeding 50% (Bekele et al., 2024).

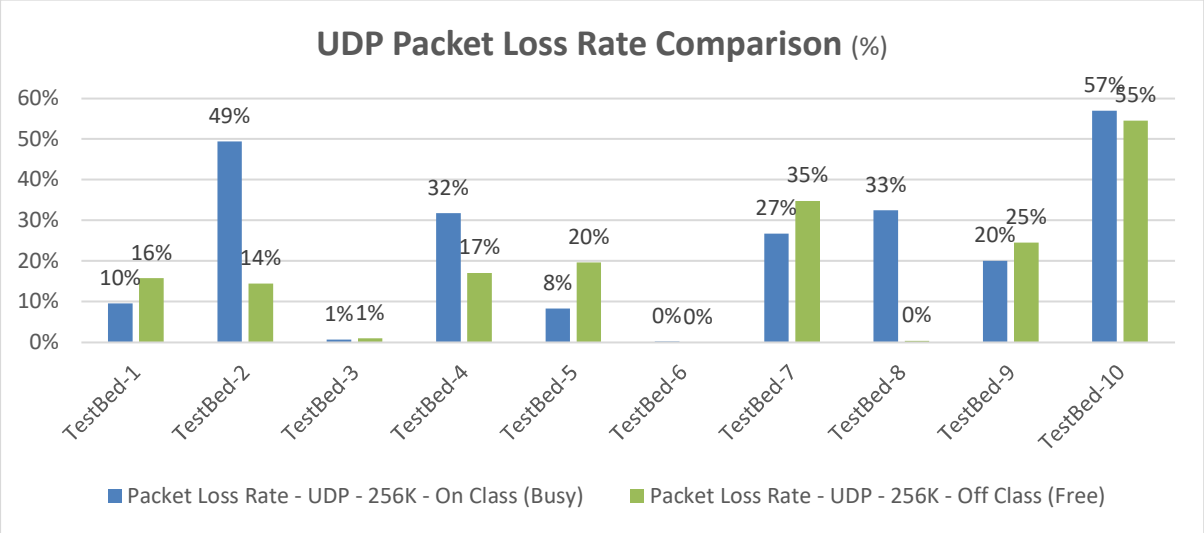

Figure 5 UDP Packet Loss Rate Comparison (%)

CONCLUSION

Driven by the curiosity about the readiness of each student's network at Advent University Indonesia and supported by a testing scenario encompassing several testing parameters, this study provides an overview of students' network conditions. In general, most classes conducted via Zoom meetings run smoothly. However, some students have difficulty receiving good video quality, as reflected in several testbeds with UDP Packet Loss Rates above 30%. Nevertheless, the audio from conversations during Zoom meetings remains clear, supported by delay values below 150 ms. Given the varying bandwidths among testbeds, it indicates that Universitas Advent Indonesia needs to consider several recommendations for the implementation of hybrid learning: 1) Ensuring that each student attending virtual meetings has adequate bandwidth (especially in the selection of ISPs (Pranata & Rizki Dewantara, 2024)), 2) Considering the use of WebRTC in optimizing virtual meeting technology (Amaral et al., 2014), 3) Creating a network prioritization scale on campus to reduce the likelihood of significant delays due to insufficient bandwidth in classroom environments.

For researchers, the subsequent steps that can be undertaken involve the development of experimental scenarios designed to simulate real-time communication between testbeds using external applications. These simulations are critical for exploring various factors that influence the quality of online learning environments. Specifically, researchers might examine the effects of different video bitrates, which can significantly impact the clarity and fluidity of video streams during virtual classes. Additionally, varying levels of background traffic will be assessed to understand how concurrent data usage affects communication quality. Another important aspect of these scenarios is optimizing network priorities. This involves creating strategies that allocate bandwidth more effectively, thereby minimizing disruptions and delays

that could hinder the learning experience. Customized bitrates for each testbed will also be determined, allowing for tailored solutions that cater to the unique characteristics of each environment.

The aim of these experimental simulations is to provide insights that can guide the effective integration of technology in hybrid learning models. By analyzing the results, educators and institutions can better understand how to leverage both on-campus networks and rural connectivity to enhance educational outcomes. This knowledge is vital in ensuring that all students have equitable access to high-quality learning experiences, regardless of their location. Through comprehensive testing and analysis, the research aims to contribute significantly to the field of educational technology.

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