Performance evaluation of VoIP analysis and simulation

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ABSTRACT

The use of technology has impacted on communication in so many ways. The advent of voice over internet protocols (VoIP) has made the world a global village where one can reach out to any part of the universe. But a challenge exists as to how to make communication and data transmission faster, the volume of traffic, bandwidth and latency in networks, that has to be transmitted between the sender and the receiver. The overall customer experience can be improved by the use of technology, which also makes it simpler to collect client information. Data packets are addressed and routed by the Internet Protocol (IP). This research aimed at deploying jitter, throughput, network traffic delay and bandwidth (JiTTraB) as a performance metrics to measure voice over internet protocols (VoIP) to measure the Quality of Service (QoS) of networks. This method prioritizes network traffic going through a router and providing acceptable service to most users in a quest to address VoIP concerns. In comparison to an existing university campus network (UCN), JiTTraB has a higher throughput, more data transmission at a given time with a minimal delay of about 0.001 seconds. However, UCN queues data with a high rate of delay before transmission. Data seem to crash considerably on the UCN due to network overload above its throughput margin and growth capacity. Thus, the proposed network design based on JiTTraB outperforms UCN in terms of transmission rate, throughput, bandwidth, delay and jitter. In addition, the JiTTraB communication network enhances QoS by performing better in close-quarters conversations than in far-quarters conversations. With an expansion reserve of 30%, the JiTTraB can handle up to 350 conversations securely while maintaining VoIP QoS standards without degrading the functionality of other network services or applications.

Keywords: Internet Protocol (IP); Voice over Internet protocol (VoIP); Throughput; Quality of service (QoS); OPNENT.

1. INTRODUCTION

Video and voice packets delivered over internet protocols (IP) are prone to certain transmission issues due to the nature of Internet Protocol networking. To conserve bandwidth and lower transmission costs, low-bit-rate codecs are employed in video and voice-over IP. Because the Internet is still mostly a best-effort network with little support for service quality, packet loss could make music or video less clear. To improve quality, manipulators must employ end-to-end strategies. Some qualitative components are lost when human speech is translated to analog electrical impulses and then digitized and compressed. Echo, jitter, and delay can all be caused by network conditions. Quality of Service (QoS) methods are required to overcome these issues. The audio signal's clarity, also known as cleanliness and crispness, is crucial. The listener must be able to identify the speaker and determine the speaker's mood [1]. Data signals are pulses or signals that carry physical data from one source to one or more destinations, typically in binary form. To be transmitted across a network, data must first be converted into electromagnetic signals. In

contrast to voice, which is often encoded using an analog format, data, and video signals are typically encoded using digital technology. In analog technology, the sound is recorded in its original form [2,3].
Voice over Internet protocol (VoIP) refers to a way of providing audio and video communications that substitute internet streaming technologies for traditional fiber optics. Video and voice-over internet protocols (VVoIP), which is what most creators and consumers of the technology are familiar with, on the other hand, is more commonly referred to as voice and video over Internet protocol by many customers and enterprises [4]. The fundamental principle of VVoIP is to transform visual and audio data into digital data, which is further compressed into packet-sized components, and then use these packets to create a data stream over the Internet. This is frequently referred to as point-to-point communication. The data stream could involve a point of origin connected to several sites of termination that are all connected to the Internet, or it could be used to transfer data back and forth between two places. Conventional technology, which employs circuits from the public switched telephone network, cannot be used [5]. VVoIP is a fantastic tool for streamlining a company's communication procedures. Implementing a VVoIP communications strategy can frequently result in considerable cost savings for telecom services as well as potentially extending the utility of conventional phones and videos much beyond the present applications that the company employs daily. However, there are a few things to take into account before getting on the VVoIP bandwagon [6].

How much bandwidth the existing Internet service provider can offer is a very important Metric in this research. As more service providers increase bandwidth to meet client demand for VVoIP, connections may occasionally be delayed or even lost, but this is becoming less of a concern. The assessment of multimedia transmission using Internet IP and QoS has received a lot of attention in recent days. Assessing how people can use their communication tools in the next stage of communication is very essential. People Utilize IP and QoS as protocols to test ability of networks to support daily teleconferences, audio, and video streaming for significant meetings, and all of the point-to-point online presentations that sales staff will deliver to potential customers. The performance of several codecs in video and VVoIP networks has been assessed using quality metrics such as bandwidth, jitter, and latency. The network operates as a local exchange for various call types and was built within several private networks, including Wire and Wi -Fi [7]. The device or software program that is specifically designed to enable voice or video communication exchange over an IP network was the target of evaluation. In [8], the study demonstrated how various queuing disciplines, such as priority queuing (PQ), first in first out (FIFO), and weighted-fair queuing (WFQ), affect application performance and resource utilization by forecasting packet loss. End-to-end latency was computed after a packet was received and voice communication was conveyed. In the end, it was demonstrated that PQ outperforms WFQ and FIFO for real-time communications like VoIP. PQ decreases end-to-end delays as well. Due to its increased packet loss, FIFO was found to be unsuitable for real-time applications. As a result, a consideration was given to real-time application factors like speech quality and resolution to determine the best queuing discipline for high packet transmission and reception with low E-E latency. Using the OPNET network simulator in [9], a complete simulation technique was described for successfully implementing VoIP. The method and outcomes established the maximum number of VoIP call that a present network can support while preserving network service quality criteria and allowing for future expansion before VoIP equipment is installed. Graphical data from a series of simulations on a segmented network that solely considered peer-to-peer calls for VoIP traffic and other relevant devices were utilized in the investigation. Several quality-of-service factors, such as mean opinion score, packet loss, jitter, and latency, which were measured and evaluated in [10] using the optimized network engineering tool (modular 14.5), had a significant impact on the performance of VoIP in WiMAX networks with better voice codec selection. Multiple voice codec systems and actual networking environments were employed in the investigation. The authors in [11] compared the performance of VoIP to digital communication on well-known applications like Skype and MSN. They assessed the effect of VoIP on the overall QoS, built a more precise topology by using new models, and considered potential VoIP delays and distortion issues as traffic loads grew. After simulating under five scenarios; VoIP conversations within a LAN, long-distance VoIP calls within a LAN, VoIP talks within a LAN with an FTP server, and VoIP calls within a WLAN with interference, it was shown that Ethernet has a more dependable and low-latency connection than Wi-Fi. Interference severely lowers the QoS of a wireless router. Additionally, jitter, ETE, and MOS are all affected negatively by long-distance VoIP. Due to this, VOIP was thought to have the potential to eventually replace the current circuit-switched phone network, despite some of its drawbacks. The study in [12] considered various indicators that gauge how the network's quality is declining, including voice data length, jitter, bandwidth, codec, packet loss, throughput, latency, and de-jitter buffer size. The analytical mathematical E-model was used to anticipate how well the current network will handle VoIP. High levels of consumer satisfaction and great voice quality were indicated by the transmission rating factor R, which was found to be 85.08 in this case. The results show that VoIP may be deployed over WLAN with perceived good speech quality, user satisfaction, small latency, and high throughput. The network was modeled and simulated using Riverbed Modeller Academic Edition [13,14]. A variety of WiMAX network and VoIP settings, including WiMAX service classes, mobility, node count, and VoIP codecs, were examined to evaluate the performance of VoIP over mobile WiMAX networks. Throughput, mean opinion score (MOS), jitter, and latency were some of the variables analyzed. OPNET Modeler was used to design and simulate several WiMAX network scenarios to get the desired results. The results showed that the UGS service class had the best performance standards for offering VoIP. Additionally, it is discovered that the G.723.1 codec uses little bandwidth while having a lower latency and higher MOS. Communication networks were analyzed on VoIP using OPNET, which projected that in the long run, the future will experience persistent widespread use of VoIP because of its numerous benefits. They looked at how VoIP quality factors like jitter, voice end-to-end delay, packet loss, and Internet QoS affected the call's quality. It was discovered that the VoIP network's quality degrades as it becomes busier. Additionally, it was discovered that VoIP quality is impacted by internet QoS, leading to the conclusion that bad internet QoS results in a higher packer discard ratio, indicating the tendency for more voice packets to be discarded, which muddles the voice message. It is possible to establish how the high packet ratio impacts other VoIP degradation variables such as jitter and end-to-end latency as well as the effect of compression on VoIP quality by contrasting three-speed codecs (G 711, G723, and G729). The simulation outcomes for the three codecs are consistent with the compression theory. Building the recommended network and evaluating the impact of various router queuing algorithms on VoIP QoS were the study's objectives in [16]. The "OPNET Modeler version 14.0" (VoIP Network) simulation tool was used to simulate local and long-distance communication; the key elements which, by the standards of the International Telecommunication Union (ITU), have an impact on VoIP QoS, such as delay, jitter, and packet loss, were computed. Several queues, including priority queue (PQ), weight fair queue (WFQ), and first-in-first-out (FIFO) were examined. It was discovered that PQ and WFQ are the best techniques for enhancing VoIP QoS. It is exciting since phone conversations can be made over data networks. IP telephony, also referred to as Voice over Internet Protocol (VoIP), is the technology used and it has lately been widely accessible. In [17], simulations of SIP-based VoIP, were used to build unified communications (UC) for the network of Mosul University. A real-time, time-sensitive communication service is voice telephony. Before the new VoIP service was launched, the IP network at Mosul University was simulated using OPNET network simulation software to make sure it was ready and able to handle this new form of traffic.

It is very clear from the foregoing literature that most of the research attempts focused on specific targets by deploying some specific metrics. For instance, the study in [8] emphasizes various queuing disciplines such as PQ, FIFO, and WFQ. In [9], the concentration was on establishing maximum number of VoIP calls that can be supported by an existing network while preserving network service quality criteria and allowing for future expansion. Many more considered one particular network. None have considered applying all the key metrics in analyzing the quality of service (QoS) of communication networks. It is believed that by applying all these key metrics, a complete communication network analysis can be conducted to inform the design of a cutting-edge network that enhances QoS. Filling this gab constitutes the major drive of this paper. The key metrics to be integrated into the design of the proposed network include jitter, throughput, traffic flow, growth capacity and bandwidth (JiTTraB).

Thus, the paper aims to deploy various software to analyze, simulate, and evaluate voice over internet protocols to characterize the Voice Performance metrics of the JiTTraB network. This is expected to enhance Quality of Service to address VoIP concerns.
The following are the contributions of paper. The main contribution of this paper to science is to propose the JiTTraB metrics and integrate same in the design of communication networks. With the JiTTraB communication network, QoS is enhanced in the following ways:

- VoIP performs better in close-quarters conversations than in far-quarters conversations.
- with an expansion reserve of 30%, the proposed network can handle up to 350 conversations securely while still maintaining VoIP QoS standards and without degrading the functionality of other network services or applications.

2. MATERIAL AND METHODS

This section describes the modeling and simulation of the work, including formulations, queuing disciplines, flowcharts, simulation parameters, and simulation. It also includes descriptions of the various software and component types that were employed. Software for flowcharts, modeling, and simulation. *3.1 Software Requirement*
The many software programs utilized for this project are described. Some of the software used are mainly for network analysis and network monitoring. The software include; Click Charts, OPNET, Cisco Packet Tracer, and Wireshark.

3.1 Opnet
A commercial simulation platform offered by Riverbed Technology is OPNET Modeler, now known as Riverbed Modeler was used. It contains a very large library of standard models, most of which are provided by the suppliers themselves. With the use of this library, practically any current network can be built on top of simulation, and network analysis can be used to determine how different scenarios and technologies would affect end-to-end behaviour. Additionally, OPNET Modeler offers a very user-friendly Integrated Development Environment (IDE) for developing devices, protocols, network processes, and algorithms throughout the communication stack [18]. OPNET was used for the simulation of the University Campus Network to which the proposed JiTTraB network is compared with reference to the performance metrics of both networks.

3.2 Cisco Packet Tracer
Cisco Systems developed Packet Tracer, a cross-platform visual simulation application that enables users to design network topologies and simulate modern computer networks was employed. Users can practice configuring Cisco routers and switches using the software's simulated command line interface. The user interface for Cisco Packet Tracer depicted in

Figure 3 is used for designing the network structure and configuring the performance metric of the project's network.

3.3 Wireshark

The Wireshark packet analyzer is open-source and free to use. In this article, the network is assessed as it is being simulated using the Wireshark analyzer to ascertain several properties. Analysis, network troubleshooting, software development, communications protocols, and education makes it user friendly.

3.4 VoIP Performance Metrics Formulation

VoIP tests capacity to communicate clearly while being background noise-free. Throughput, jitter, MOS, delay or latency, and error rate are the main parameters that have an impact on VoIP speech quality [19,20,21].

3.4.1 Throughput

The throughput is used to measure the quantity of packets transmitted in a network. The transmission rate can also be used to estimate the throughput of a network. As indicated in [22], the unit of measurement is packets per second or per time slot. Equation 1 can be applied to determine throughput.

$$\text{Throughput} = \frac{\text{Total bytes received} \bullet s(bit)}{\text{End time}(s) - \text{start time}(s)} \quad (1)$$

3.4.2 Mean opinion score (MOS)

The MOS is the most popular gauge of voice quality [23,24]. MOS is a useful benchmark for network evaluation, benchmarking, tweaking, and monitoring due to the correlation between audio performance parameters and a quality score. One to five excellent make up the MOS value range. With the use of the e-model, a computer model is used in transmission planning. The MOS is calculated using a non-linear mapping from the R-factor. Equation (2) shows the ITU-T guideline [25].

$$\text{MOS} = 1 + 0.35 \times R + 7 \times 10 - 6\left[R(R-60)(100-R)\right] \quad (2)$$

Where: $R = 100 - I_s - I_e - I_d + A$.

I_d: represents the delay-related disability, in particular mouth-to-ear delay. I_s the result of voice signal impairments, I_e, the impairments brought on by different kinds of losses brought on by networks and codes.

3.4.3 Jitter

By comparing the end-to-end latency of two successive packets, it is possible to identify network jitter, which is brought on by traffic, route changes, queuing, etc. The system's latency variations are then displayed after calculating the jitter value. A network's effectiveness is established, as well as its dependability and consistency [21,26] Jitter is the signed maximum fluctuation in the one-way delays of the packets over a predetermined period. Equation 3 can be used to calculate it:

$$\text{Jitter} = \text{Max}1 \le i \le n\left\{\left[t'(n) - t'(n-1)\right] - \left[t(n) - t(n-1)\right]\right\} \quad (3)$$

Where: t(i) and t'(i) are, respectively, the transmission time at the transmitter and the signal reception time.

3.4.4 First-in-first-out (FIFO)

In this case, the first packet added to the queue will be broadcast first. Due to the limited capacity of the router's buffer, a packet will be dropped when the buffer is full. This technique doesn't provide packet transmission in a network any priority. Real-time applications are not suited for this queuing method [27].

3.4.4.1 Priority queuing (PQ) and Weighted-fair queuing (WFQ)

This type of FIFO queuing assigns a priority indicator to each packet and tags it in the ToS field. Numerous priority-classed FIFO queues are used in this queueing technique at routers. The buffer sorts the packets as they enter it. Initially transmitted will be the packet with the greatest priority. Real-time applications like VoIP are best suited for it. Normal packets will have to wait a long time in the buffer since higher priority packets will be delivered first, which creates a famine condition [27].

Every flow is given a weight by Weighted-Fair Queuing (WFQ). The weight is identified by the ToS field. This blends PQ and FQ. Packets are weighted in this queuing technique such that those with less capacity are given more priority. By pushing traffic ahead of the queue with larger, slower-moving packets that are sent more quickly, WFQ enables real-time interactive applications. Sometimes it is ineffective because it makes the network more congested [28,29].

3.5 VoIP Deployment

An 8-step technique is provided in the flowchart below for a successful VoIP setup [30]. The first four stages can be carried out concurrently. Step 5 must be completed first, requiring an upfront change to the current network, before the simulation in step 7 can begin. You can carry out steps 6 and 7 in order. The deployment of the pilot is the last phase.

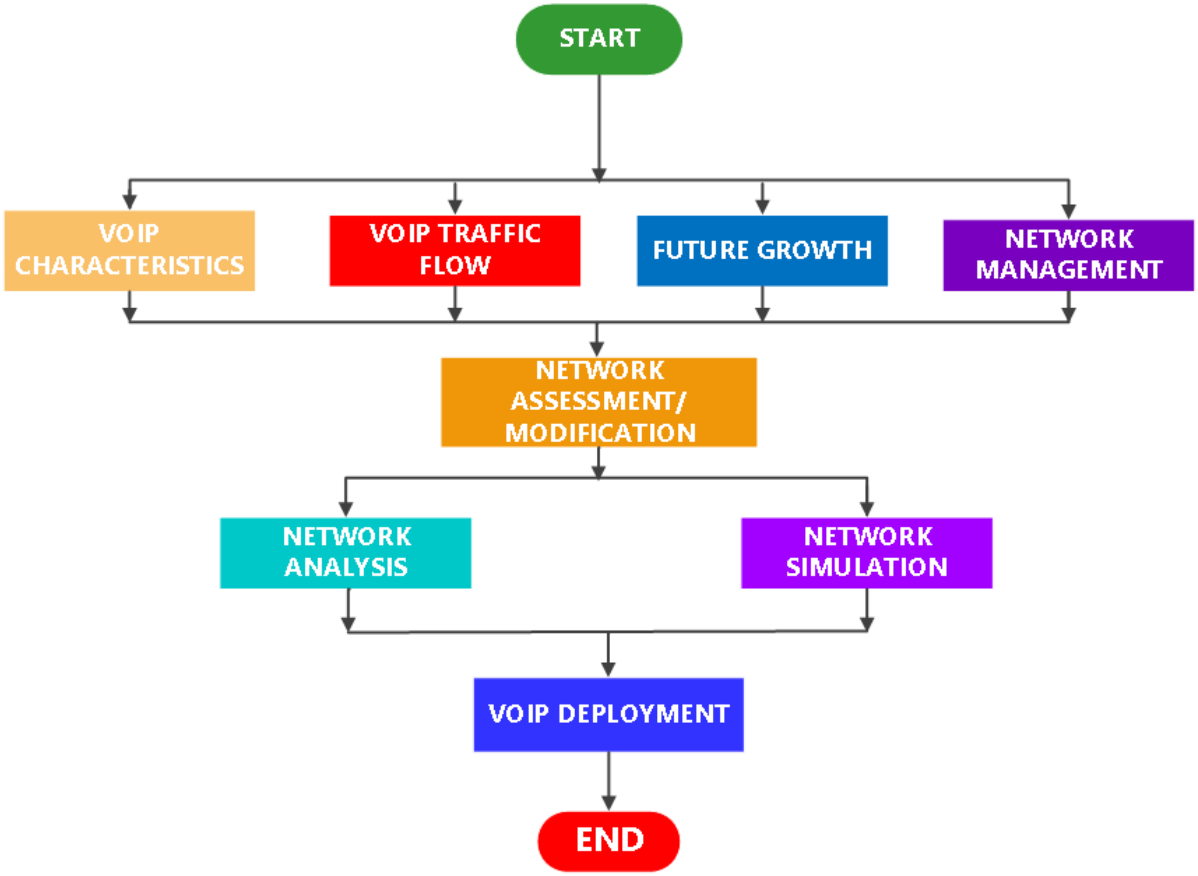


Figure 1: The VoIP Deployment Flowchart.

3.5.1 VoIP traffic haracteristics, conditions and assumptions

In implementing a new network service, such as VoIP, it is necessary to determine the type of traffic as the flowchart shows in Figure 1, the QoS specifications, and any extra components or devices that can be implemented. This study makes the simple assumption that all VoIP calls are one-on-one communications without call conferencing. In this research a performance characteristic for encoding and decoding the side of PCM channels applicable to voice frequencies (G.714) is deployed. It requires a 150 ms maximum end-to-end total one-way packet delay for VoIP applications [18,31,32]. It was decided to put up with delays of up to 200 ms. In this article, delay may result from the following: (i) delays brought on by network propagation, transmission, and queuing; (ii) delays induced by encoding, compression, and packetization at the sender; and (iii) delays brought on by buffering, decompression, depacketization, decoding, and playback at the receiver. In this article a VoIP call requires a 64 kbps of bandwidth.

Using the preset voice application is the method used for simulating the VoIP communication in the Cisco. The Cisco is as a series of phases, application that is essentially used in collection of tasks. There are two endpoints involved in each step, and the traffic behavior is adjustable depending on the volume of traffic. In this design the begging and how long a job will take is specified [16].

3.5.2 Growth capacity and network measurements

The network's anticipated growth is fully taken into account when determining the expansion capacity or future growth factor required for the network to accomplish a given goal. Our network’s capacity is expected to expand by 20% more for future expansion. To keep things simple, switched-Ethernet lines, routers, and switchable switches all have equal amounts of network resources. But in practice, this proportion might vary depending on the presence or absence of the desired expansion capacity of each network resource. The network resources are reserved in advance of delivering the new service in this simulation technique, and only the remaining capacity is used to test the network's ability to handle the new service [33].

To establish the current traffic load, measurements were made. This crucial step might potentially have an impact on the simulation's findings. To do network measurements, a variety of commercial and open-source tools **can be** used. There open-source measurement and analytical tools such as Wireshark, is used to monitor the traffic load on network links, and Spanning Tree Protocol (STG) is used when multiple paths exist for configuration [34]. For a thorough analysis, network measurements were collected over an extended period. Measurements is taken over a week or many days. To guarantee sufficient quality of service (QoS), particularly during peak hours, network traffic or consumption must be taken into account depending on usage and the time of the day. [35].

3.5.3 Network assessment/modification and analysis

In this work the network is evaluated in terms of the volume of traffic that is now present and the specifications of the anticipated new service. Immediate network changes include but are not limited to, PC upgrades, the inclusion of new servers or devices, and the resizing of connections that are often used can affect network modification. As a result, general upgrading, topological changes are maintained to the barest minimum unless the modification is minimum.

Two significant considerations impose limitations on VoIP. The available bandwidth and end-to-end delay [30]. Depending on the network, the most crucial factor in determining the number of calls that can be accommodated by the network is the available bandwidth and the latency. These two indications impose a limit on the real number of VoIP conversations that the network can handle and manage.

3.4 Bandwidth bottleneck and delay analysis

Bandwidth bottleneck analysis, which identifies the node or connection with the smallest available bandwidth for each route among number network nodes and connections, is a crucial first step in identifying the network element be it a node or a link that restricts the number of VoIP calls that the current network can support [36].

At this time, VoIP packet delays up to 150 ms are supported by the network. The quantity of VoIP calls that may be processed simultaneously is limited by this latency. In this work, it was tried to figure out the maximum number of calls the existing network can handle while still maintaining VoIP QoS. As the network call capacity slowly rises, monitoring the VoIP latency limit or threshold might assist in achieving this. The maximum number of calls may be known if the end-to-end latency, including network delay, is greater than 150 ms.

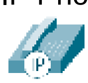
3.6 Simulation Scenario and Modeling Network

The purpose of the simulation is to validate the analytical results regarding support for VoIP calls. The commercially available network elements are extensively modeled in OPNET

Modeler, which also offers a variety of real-world network design options. This enhances the realism of the network environment simulation. OPNET also includes a sizable library of network protocols and models, a graphical user interface, statistics, and graphs.

An Ethernet workstation serves as the paradigm for the VoIP gateway. Ethernet server architecture is used by commercial servers. A 100 Base-T link connects each component of the network. Figure 2 displays the network topology.

Table 1: Components Specification

S/N	COMPONENTS	DESCRIPTION	QUANTITY
1	2811 Integrated Services Router (ISR) Router 	Enhanced Network Modules are available with two slots for Advanced Integration Modules (AIMs), Two fixed 10/100 (100BASE-TX) Ethernet ports, four integrated High-Speed WAN Interface Card (HWIC) slots, and two slots each for Voice/WAN Interface Cards (VWICs), Voice Interface Cards (VICs), and WAN Interface Cards (WICs).	2
2	Switch 	A switch from the Cisco Catalyst 2960 Series is the WS-C2960-24TT-L. Cisco Catalyst 2960 Series switches offer voice, video, data, and extremely secure access. Additionally, they offer scalable management to meet changing business requirements.	6
3	PC 	One 2.4 GHz wireless interface is provided by the WMP300N module, which can be used to connect to wireless networks. The module is compatible with Ethernet-based LAN access protocols.	12
4	IP Phone 	A full-featured IP phone designed particularly for manager and executive needs is a prominent component in the IP Phone line. To help a user, navigate the call features and functions, it offers four interactive soft keys and six programmable line/feature buttons. Duplex speakerphone, handset, and headset audio controls.	6

The components are connected as shown in Figure 2 to simulate the VoIP network in other to analyze the network traffic together with the traffic characteristics.

3.8 Simulation Procedure

Figure 7 displays the simulation model for the present network being evaluated. The offered organization's network simulation model is an exact reproduction of the actual network. However, the necessary hardware the Cisco 2811 ISR router and the catalyst 2950 is not yet available. Ethernet switches were readily accessible in our network. The VoIP gateway was created as an Ethernet workstation since gathering statistics inside the corporate network is its primary goal. Ethernet servers serve as a model for commercial servers. Each part of the network is connected via a 100Base-T connection.

To replicate the actions of LAN users, floor LANs are modeled as subnets that surround an Ethernet switch and three particular Ethernet workstations. The floor's background traffic is produced by one of these workstations, while VoIP sessions are conducted on the other two. Network Class A, Network Class B, and Network Class C are some examples of the labels used to identify the Ethernet workstations. One way to send VoIP calls is using Network Class A. VoIP calls can be received at Network Class C as a sink. At Network Class B, background traffic starts and ends. Another finding is that floor multimedia PCs and IP phones don't exactly match floor LANs. If a model is created with such precise floor network setups, the simulation will be virtually entirely manual. This is so that two jobs can be accomplished every time a new VoIP call (or group of calls) is added: first, add distinct PCs with various profiles and settings, and second, execute the simulation. After every simulation run, this must be manually done, and the results must be reviewed. Since it is programmed to produce three calls automatically every three seconds, the simulation technique is automated. The effectiveness of internal nodes and links within the core network is not significantly impacted by the technique or model.

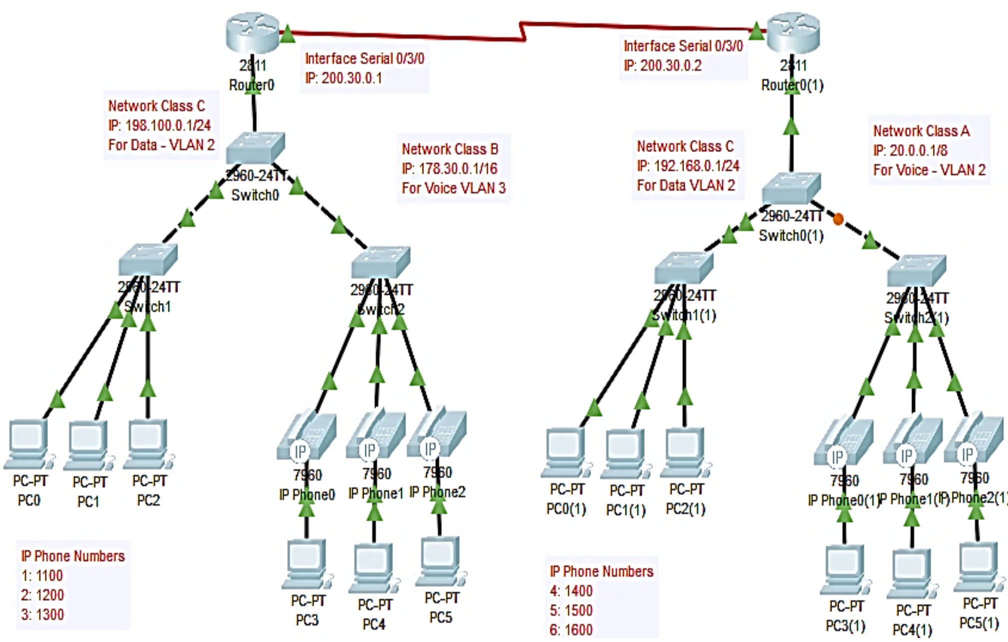


Figure 2: The VoIP Deployment.

From the diagram above, the structure under 2811 Router0 represents a network system of an Area(A) whereas the structure under 2811 Router0(1) also represents another Area(B). Messages/Data are sent using the PC-PT whilst calls are made using the IP Phones thus communication initiated is depicted in Figure 8 as the green arrows show the flow from PC-PT PC4 to PC-PT PC0. Whereas the yellow-colored message icon shows a pending message to be sent.

The network flow begins from the PCs or IP Phones. When messages/calls are initiated, the data acquired is sent to the 2960-24TT Switches to provide fast connectivity thereby enhancing switching services, advance security, IP communications, wireless networking and scalable management. The switches then check for the destination of the Data when it is within the Area it is managing. Otherwise, it carries it on to the Router0 to be sent to another Area to be checked and delivered to its final destination.

3. RESULTS AND DISCUSSION

The result of the simulated network is shown in Figure 3. The network works perfectly when a limited number of individuals are on the network within the same community as seen in the right bottom corner of the interface. But in Figure 3 communication over internet protocol will be carried from one community to another on a loaded network.

Considering the diagrams in Figure 3 and Figure 5, a message sent from PC-PT PC4 seemed to have failed to reach the desired destination PC-PT PC5(1) due to a latency in the network flow from 2960-24TT Switch0(1) to 2960-24TT Switch2(1) indicated by an orange point/dot. As a result of the failed transmission of data the number 0f packets was reduced, successfully transferring a measure of data to destination at 132kbps within a bandwidth speed of 15 Mbps.

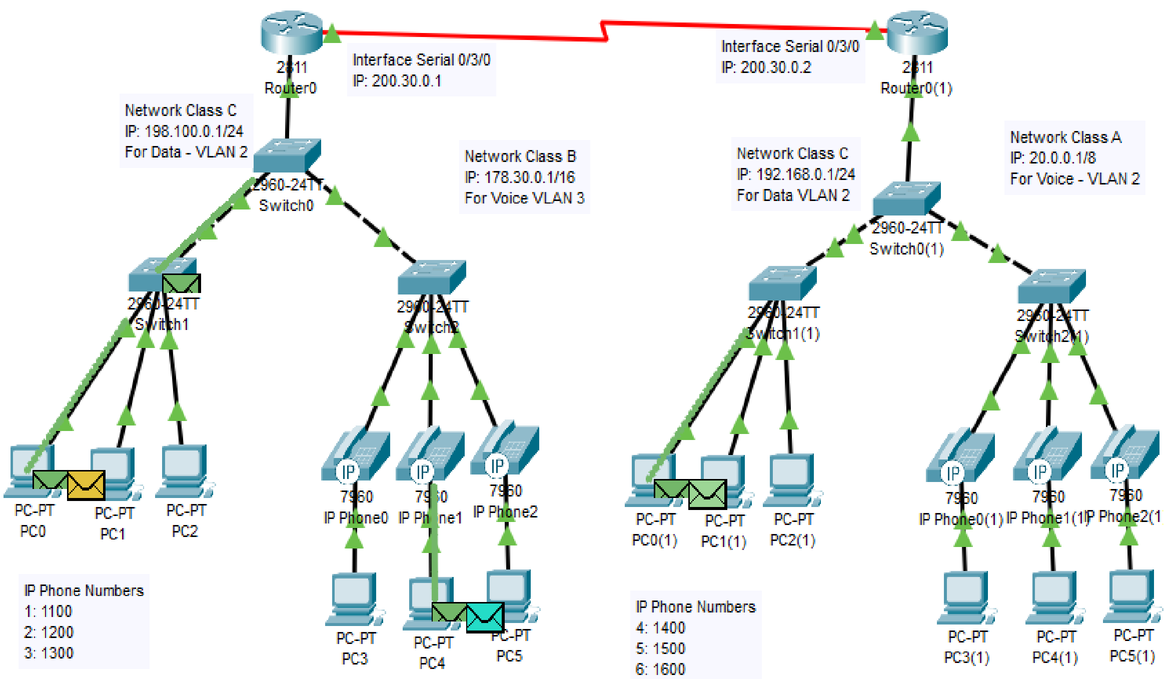


Figure 3: Simulated Network Scenario 1.

As shown in Figures 3 and 4, the communication over a loaded network was quite delayed and failed to be transmitted to the next community. This results in the transmitted data being lost on the network. As the simulation was ongoing the graphs were generated using Wireshark. And is displayed as follows;

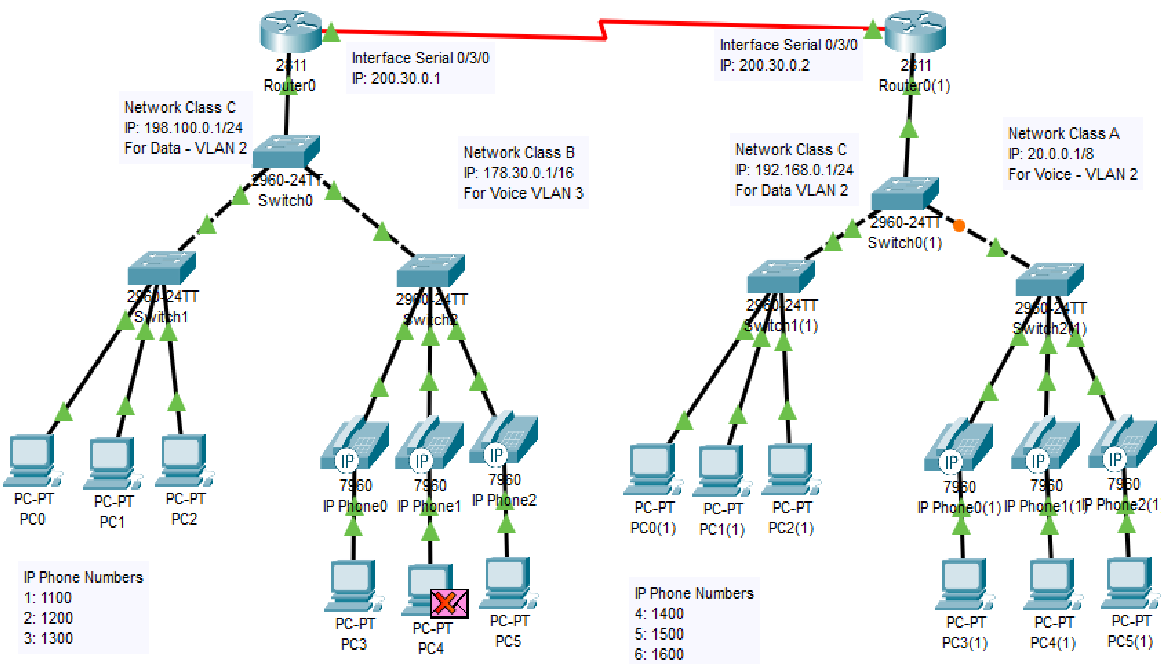


Figure 4: Simulated Network Scenario 2b.

After completing the network and simulating it there seemed to be traffic on the network when more than one individual operates at a time. The network is operating efficiently when there is approximately the same quantity of traffic sent across all queue disciplines and no packets are dropped (Figure 5).

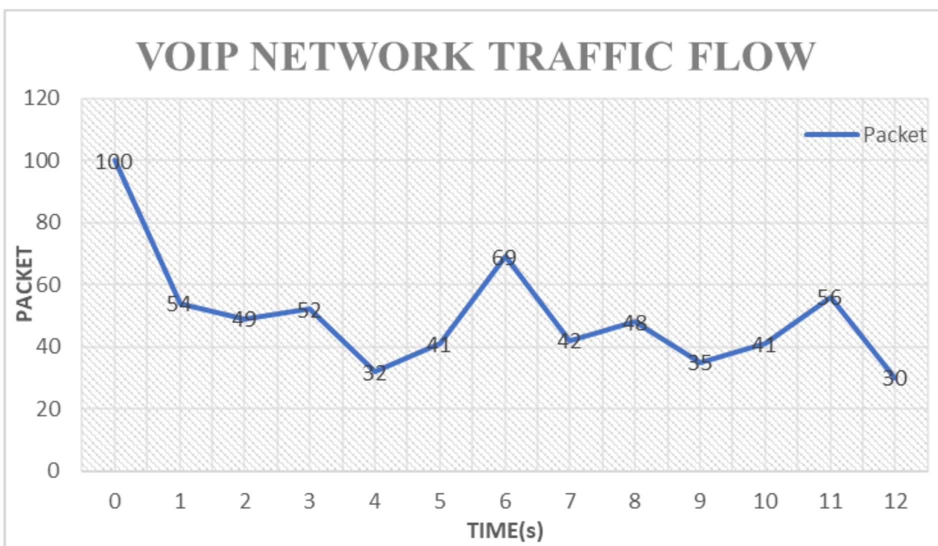


Figure 5: VoIP Traffic.

The throughput depicted the number of packet quantities transmitted per time slot but since it is over the internet time and rate of size are considered. The network started transmitting 590 packets at the initial start of the simulation which gradually increased to 1180 packets per second as the number of operators on the network increased. The flow of packets from the sender node to the destination node is shown in Figure 7's throughput. How many packets the destination may receive in a specific amount of time is indicated by the maximum throughput.

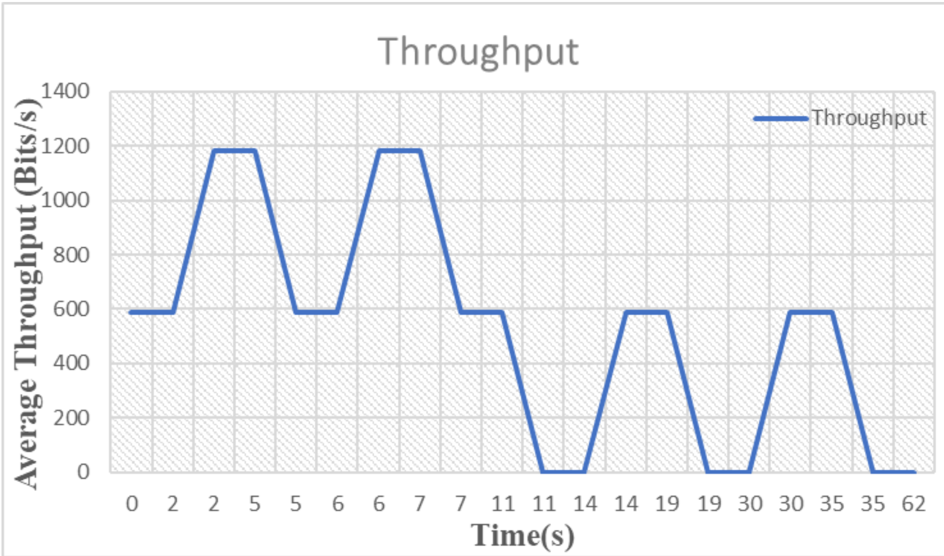


Figure 6: Simulated Network Throughput.

The figure (7) displays a delay of 22ms for the data that needs to be filtered before transmission. this means that it takes some internet protocols about 22 milli-seconds to filter and transport data, which can occasionally cause jitters, delays, and data losses. The filtration of data

The proposed designed network is compared to another network of mass usage. The alternate network being compared here was designed for a major campus usage comprising the entire student and staff body of the school.

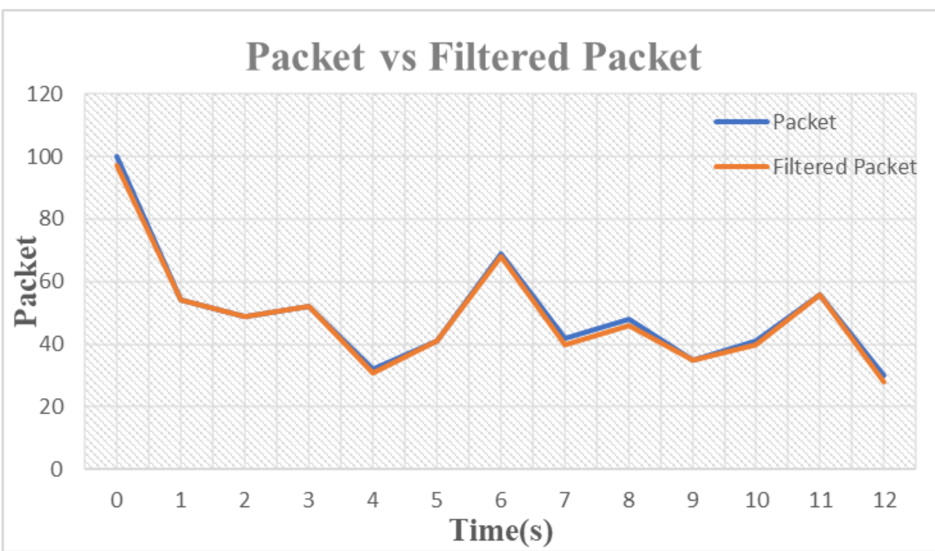


Figure 7: Packets and Filtered Packet End-to-End Delay.

As shown in the Fig. 8, UCN is the existing University Campus network design, and our proposed network design is JiTTraB. As the graph depicts JiTTraB has a higher transmission rate as compared to the UCN. This is due to the margin of throughput and growth capacity allowed on the JiTTraB being greater than that of UCN.

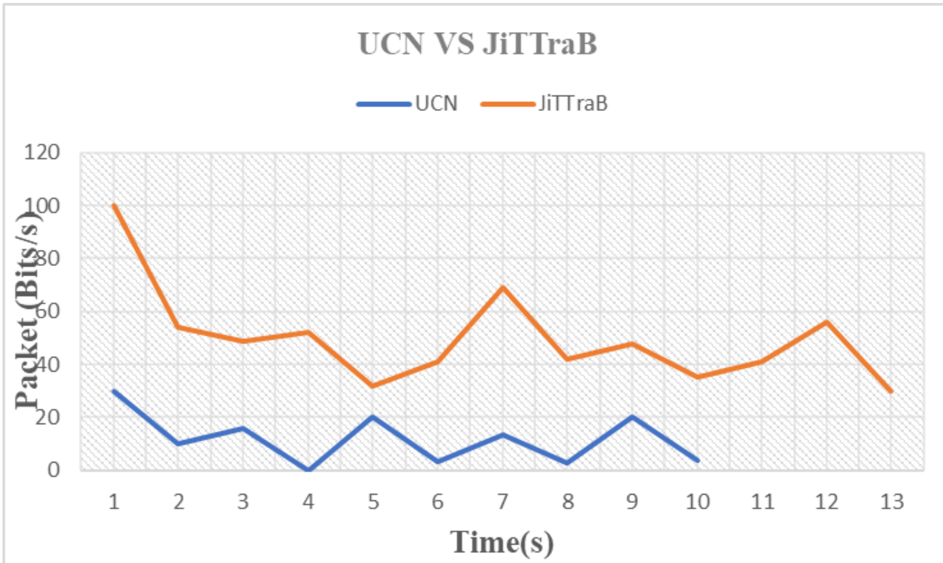


Figure 8: UCN vs ASEN

As depicted in Fig. 9, the throughput of the proposed JiTTraB is greater than that of UCN due to the data capacity which is transmitted over the network at the given time. JiTTraB has a greater throughput due to the consideration of increasing the capacity of the network, thus efficiently giving rise to a bandwidth range of 20 to 190 kbps per connectivity and a measure of data successfully transferred to the destination between 90 kbps to 156 kbps at a bandwidth speed of 5 to 25 Mbps. At 1180 b/s, the transmission of data begun to reduce drastically to 590 b/s and maintained for a second due to the reduction of data being sent over the network by users and then increased again due to an increase in the number of data needed for transmission. This implies that throughput is indicated by the number of users at a point in time against the number of data sent.

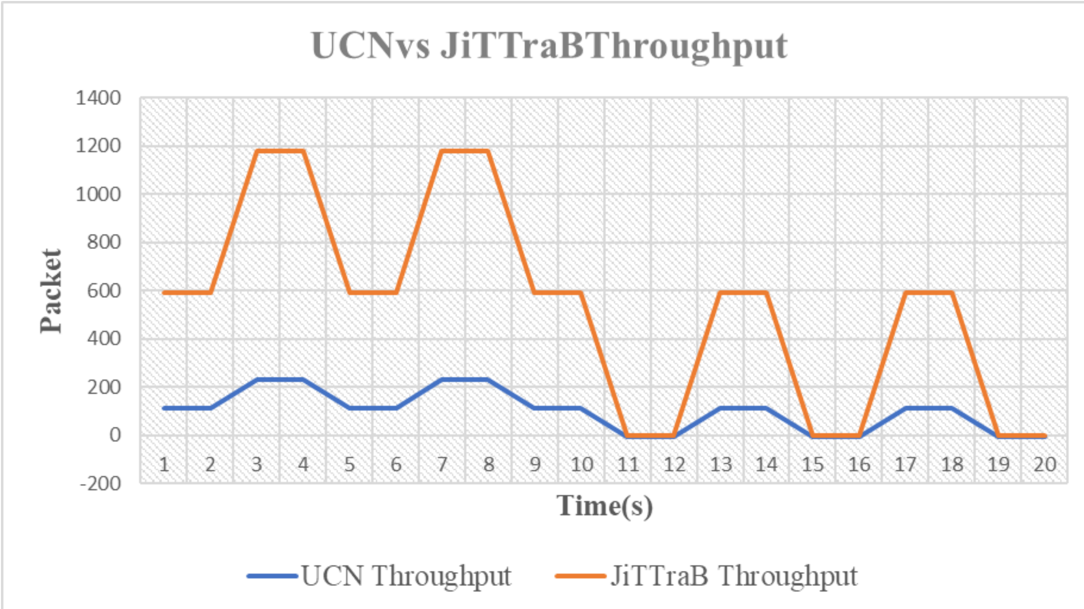


Figure 9: UCN vs ASEN Throughput.

As seen in Figure 10 below UCN has more data clashes (losses in packet transfers) than JiTTraB.

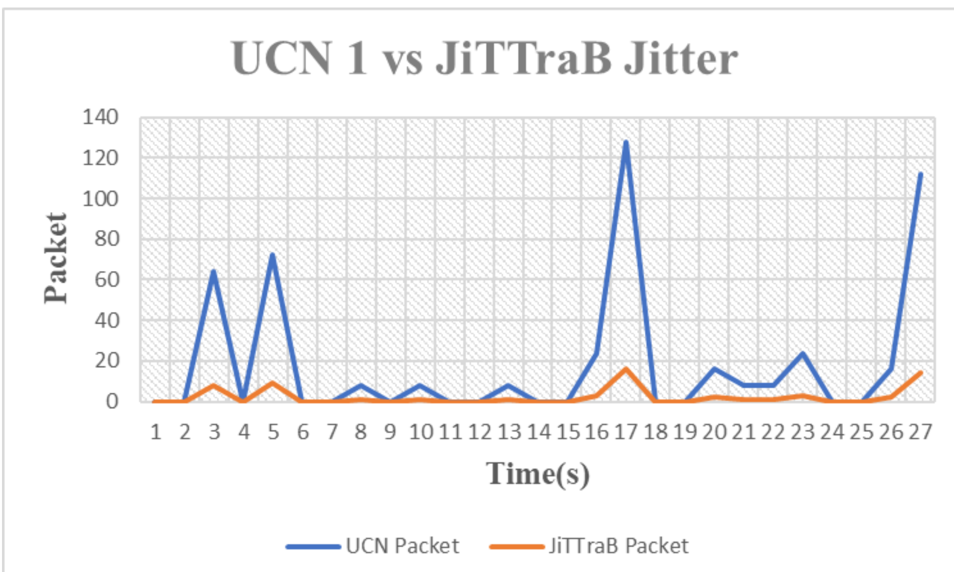


Figure 10: UCN vs ASEN Jitter.

After 15 seconds of, simulation period it was realized that the UCN jitter rises to a 128-packet data clash due to the overload of the network as emanating from excess capacity. As more data are transferred on the network, it begins to crash into other networks, which may also result in data loss whiles the JiTTraB maintained its latency rate between 0 and 16 kbps thus incurring very minimal loss. Instead of the receiver hearing "Hello" he would rather receive a prolonged word "H...e…l…l…oo". Theoretically, this implies that a network's efficiency is also based on the level of measure for its latency in relation to data transferred.

The on the simulation results, the following network decision can be supported:
With a 30% expansion reserve, the current network can accommodate up to 350 conversations securely while still maintaining VoIP QoS standards and without degrading the functionality of other network services or applications. All network resources are maintained with a 40% safety growth factor. The throughput of the network rather than the latency is what sets a limit on its ability to serve VoIP. This is a result of the tiny size and low number of intermediary nodes in the present network under consideration. If the LAN or WAN was large-scale, the network latency bound might take control.

4. CONCLUSION

VoIP parameters are affected by factors such as throughput, jitter, network traffics, and delays. These factors affect Voice over Internet Protocol causing loss of data and slowing the transmission of data over networks. This work sort to analyze, simulate and evaluate VoIP networks using Cisco Packet Tracer and Wireshark. The core aim is to integrate jitter, network traffic, throughput and bandwidth into a communication network to enhance Quality of Service to address VoIP concerns. The outcome is the proposed JiTTraB network. The simulation was carried out by sending data over TCP/IP adapters which resulted in packet-switching over telephone networks independently, analyzing and evaluating various factors that affect the performance metrics of VoIP network. On comparing the proposed network (JiTTraB) to an already existing network (UCN), it was noted that the JiTTraB surpasses UCN in terms of transmission, indicating a higher margin of throughput and growth capacity. In considering an increase in capacity, JiTTraB has a higher throughput, thus more data is transmitted at a given time with a minimal delay of 0.001 seconds, while UCN queues data with a high rate of delay before transmission. Data seem to crash considerably on the UCN due to network overload above its throughput margin and growth capacity as compared to JiTTraB. In terms of percentage, it can be said that JiTTraB outperforms UCN by 30% due to its supremacy in most conditions. It is however significant to note that poor weather conditions can affect the overall performance of the network overtime which either increases delay or loss of data.

COMPETING INTERESTS

Authors have declared that no competing interests exist

AUTHORS' CONTRIBUTIONS

Authors WA and FK conceived and designed the study and performed the analysis. Authors JAA and DM managed the data, methodology and performed the simulation. Authors WA and JAA managed the literature searches, edited the work and wrote the first draft of the manuscript.

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