Research Article



# **Evaluating the Performances of Digital Data Transmission Links on a Client-Server Network Model**

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Received: 01/Mar/2024; Accepted: 03/Apr/2024; Published: 30/Apr/2024

*Abstract*— Having smooth communication and high-quality performance over IP networks in the real world has been the goal of all businesses and organizations to enhance their daily business operations through having easy access to resources and data over the network without any obstructions. Therefore, Riverbed modeler software, also known as OPNET, was used in this paper to simulate a client-server model communication over an IP network as well as evaluate the performances of the three digital data transmission links (also known as data rate links) in three provided scenarios using some selected statistics which were used for the analysis of the obtained results across all the Scenarios. T1 link was used for the first scenario in communicating among all the offices (Manchester, Sydney, Chicago, and the Call Center (Nigeria)) and shows a non-optimized network performance as there is the presence of clogs (delay) in the network. The T1 link was replaced with T3 links, giving better network performance as it could transmit packets. Although network delay and packet delivery to the assigned destination were reduced, the T3 link was replaced with the OCI link to obtain better results in transmitting network packets.

Keywords- Network, digital data transmission links, Riverbed modeler, network packets, network performance

# 1. Introduction

Networking is created for connecting computing devices to flow and share resources and data. It enables communication between devices, allowing them to exchange data, access the Internet, and collaborate on various tasks. These connected devices utilize communication protocols to send information over physical or wireless technologies [1]. In this era, many organizations adopt network technologies over the Internet in the transfer and delivery of their services, and this has resulted in a high rate of data flow over the network every time and everywhere.

Data is transferred through wired or wireless media-connected nodes in a computer network. Examples of network nodes used in creating, routing, and receiving data include desktops, phones, and servers [2]. Regardless of physical connections, networked devices can communicate. Computer networks facilitate various activities such as web browsing, file and program sharing, printing, faxing, email/instant messaging, voice conferencing, and video conferencing. Computer networks exhibit diverse topologies, signal transmission mediums, traffic management algorithms, and objectives [2]

In transferring data or packets from one endpoint to another over the IP network, several network protocols are used to

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establish the connection between the nodes. These include but are not limited to Ethernet protocol, which established a new standard that enables 100 Mbps for faster data transfer rates. Local Talk: Apple Computer, Inc. developed Local Talk for Macs alone. Local Talk transmits data via CSMA/CA. Token **Ring:** During the mid-1980s, IBM introduced the Token Ring protocol, employing token-passing for access. In a Token Ring network, computers are interconnected where the signal circulates the ring, node to node, with a single digital token passing from one computer to the next in a circular fashion. FDDI (Fibre Distributed Data Interface) often establishes long-distance connections between multiple local area networks. Token-passing is utilized for access in FDDI networks, which employ a physical topology of two rings. ATM: Asynchronous Transfer Mode (ATM) enables data transmission rates of 155 Mbps and beyond. With ATM, data is consistently transmitted in packets of identical sizes, unlike other protocols that may vary in packet sizes. ATM supports access to various media types, such as videos, audio CDs, and images. Its star topology is compatible with both fiber optic and twisted pair cabling. [2]

For effective data communication over the network, four main characteristics are considered: accuracy (to deliver data accurately without any error or being corrupted during transmission), delivery (the delivery of data to the assigned

destination and user), timeliness (delivering audio and video data in a real-time manner without delay) and jitter (deals with the variation in the arrival time of packets) [3]. Uneven jitter may probably affect the timely transmission of data [3].

The listed data communication characteristics are being evaluated using the data link rate, the volume of data transmitted between two network points within a specific timeframe [4]. This concept holds significant importance in contemporary business networking, as higher transfer rates enable networks to support intricate tasks like online streaming. Familiarity with data transfer rates could enhance the business network's performance [4]. Therefore, this paper focuses on evaluating the performances of the selected digital data transmission links by designing and simulating a clientserver network model in riverbed modeler to generate and analyze their performance results and determine the data link among the three and also suggest further works on the adoption mote advanced data links for better network performance.

This research paper is organized into sections: section 1 contains the introduction of networks and their protocols. Section 2 contains related work on network overview, its types, and its network architecture. Section 3 includes the subnetting aspect of the network. Section 4 contains the research methodology, where a detailed explanation of the procedures involved in designing and simulating the proposed network model is clearly stated. Section 5 contains a discussion of the generated results from the simulated network, and Section 6 concludes the research work with future recommendations.

# 2. Related works

# 2.1 Network Overview

A network has been explained to comprise two or more computers that are interconnected to ensure the sharing of resources such as printers, exchange files, applications, software, or electronic communications, and these computers on the network can be connected via cables, telephones, radio waves, satellite or infrared light beams [5]. The main benefits of networking can be described clearly in terms of its security, efficiency, manageability, and cost-effectiveness as it helps in communication between users in an extensive range [5].

Understanding the network infrastructure is crucial for comprehending how data is transmitted and communicated within a network. These devices include but are not limited to, computers, hubs, switches, and routers, as they play an essential role in transmitting information or data from one endpoint to another using different technologies such as radio waves, wires, or wireless [6].

#### 2.2 Network topology

Network devices necessitate connections to various topologies through transmission media such as twisted pairs and optical fiber [7]. Multiple network topologies exist, each accompanied by distinct advantages and disadvantages. These include:

**Star Network:** Facilitates the easy addition of new nodes; however, if the central node fails, the network communication is compromised.

**Bus Network:** Known for its easy installation and minimal cabling, but the failure of the bus adversely affects the entire network communication.

**Ring Network:** Requires fewer cables, yet adding new nodes is troublesome, necessitating the interruption of the original "ring" before forming a "new ring."

**Tree Network:** Enables quick connection of multiple star networks and layered expansion; however, higher-level nodes may escalate network problems.

**Full Mesh Network:** Exhibits high reliability and communication efficiency but demands numerous physical ports and interconnecting cables, leading to high costs and complex expansion.

**Partial Mesh Network:** Boasts lower costs than a full mesh network; nevertheless, it compromises reliability.

**Hybrid Network:** Combines star network characteristics for easy node addition and traffic monitoring in the center; yet, redundancy in equipment and cables increases costs significantly [7].

#### 2.3 Types of Network

In the discussion of network types, as outlined by [8], networks are classified based on geographical coverage or users. As shown in Table 1, the geographical classification comprises of Local Area Network (LAN), Wide Area Network (WAN, and Metropolitan Area Network (MAN).

LAN is described as the type of geographical-based network that typically connects terminal computers nearby, covering a few square kilometers, that is, it is used in facilitating local communication within a few covered areas. WAN is another type of geographically based network which covers a wider range of locations than LAN from tens to thousands of kilometers. Lastly, MAN covers a wider range of locations than the first two types. It serves as a larger-scale LAN, MANs require a higher cost but offer faster transmission.

The second type of network classification is the user-based network which consists of private and public networks. The private network is designed to be limited and can only be accessed by authorized users of the company while the public network on the other hand can be accessed by anyone who pays as directed by the telecom company such as the internet

			Ta	able 1: Types of netwo	ork	
NETWORK TYP	E					
Geographical coverage-based network	Local A Network (LAN)	Area	Wide Area Ne	twork (WAN)		Metropolitan Area Network (MAN)
	Description: L	ANs	Description:	Encompassing	geographical	Description: serving as a larger-scale

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	typically connect terminal computers nearby, covering a few square kilometers. <b>Function:</b> Facilitates local communication within a defined area	regions ranging from tens to thousands o kilometers, WANs connect multiple cities countries, or continents. <b>Function:</b> supports long-distance communication, forming international remote networks	<ul><li>faster transmission.</li><li>Function: enhances the transmission media</li></ul>
User-based network	Public	Priv	
	telecom companies ar directed by the telecom	nd are accessible to anyone who pays as networks are large-scale and open to the chesic busis mak	cription: A department develops private works to cater to the specific business needs of a pany. They are designed to be restricted, viding access only to authorized employees of organization. aracteristics: These networks are particularly gned to cater to the internal needs of the ness, prioritizing them above all others and ing their design meet the exclusive demands of company.

#### 2.4 Network Architecture

Understanding the interconnection of network services and devices is crucial for meeting the connectivity requirements of client devices and applications [9]. Two prevalent types of network architecture are peer-to-peer (P2P) networks and client/server (tiered) networks [9].

#### Peer-to-Peer (P2P) Network:

**Description:** In P2P networks, tasks are distributed among all members (users) without the reliance on a central server controlling network activity. Each system runs its software to facilitate communication among all computers. **Function:** Primarily used for file-sharing purposes.

#### Client/server (tiered) Network

**Description:** In client/server networks, network clients or workstations (hardware devices enabling end-users to access data and services on the server) request resources or services from the network. One or more network servers oversee and deliver these resources or services [9].

**Function**: Centralized management of resources and services.

According to a comprehensive literature review, clientserver networks exhibit superior performance to peer-to-peer networks in response time, cost, scalability, security, traceability, and privacy [10]. The client-server model is favored for its simplicity, policy adherence, industry support, skillset readiness, and economies of scale. Despite these advantages, peer-to-peer networks foster a collaborative culture [10].

In summary, the choice between peer-to-peer and client/server network architectures involves a trade-off between collaborative culture enhancement and the technical and operational advantages the client/server model offers.

#### 2.5 Several digital data transmission systems

Radio Link Data Rates were employed to establish dynamic network conditions within Tactical Networks [11]. This approach was instrumental in quantifying fluctuations in data rates by considering metrics like latency and jitter in the network. The constructed model incorporated diverse functions, including the combination of patterns, transformation from one pattern to another, skipping between patterns, and creating loops within multiple patterns of change [11].

SIC-aware MAC (Media Access Control) protocol was introduced, focusing on incorporating concurrent transmissions and optimizing channel access strategies at the MAC layer [12]. The primary goal was to mitigate interference, whether it be of the same technology or crosstechnology, stemming from excessive channel contentions. The study also involved an analysis utilizing MIMO (Multiple Input, Multiple Output) and MU-MIMO (Multi-User-Multiple Input, Multiple Output) links with Signal Improvement Capability (SIC).

Another approach adopted was a high-precision time synchronization method in the V2X system [13]. This approach was adopted to address the critical requirement for clock synchronization among V2X device nodes, ensuring minimal errors in terms of time or frequency within each network node device. The methodology involved the fusion of multiple clock sources, incorporating three service modes: GNSS, cellular data, and local. Additionally, The method took into account the prioritization of clock sources and factored in the transmission time delay related to different interactions within the time service process, covering both hardware and software aspects. [13].

Digital data performance transmission for a fiber-radio system was evaluated using VPI photonics software for the simulation of the transmission of digital signal through an optical link of 25km of single-mode standard fiber(SM-SF) and MATLAB software was used to emulate a wireless channel highlighting four phenomena to this channel which include: multipath and slow fading, co-channel interference, and additive white Gaussian noise (AWGN) [14].

A measurement analysis and performance evaluation of mobile broadband cellular networks was conducted [15]. The data measurements were done in two scenarios: outdoor and indoor environments. The results obtained from the outdoor experiment show that the maximum average throughput with downlink and uplink data rates is 14.3 and 7.1 Mbps, respectively. Also, the minimum average ping and loss are 36.5ms and 0.14, respectively, for all Mobile Network Operators.

# 3. Subnetting

Subnetting is the process of breaking a larger network into smaller, logical segments [16]. Subnetting involves the logical partitioning of an IP (Internet Protocol) network, the medium for transmitting data between computers (hosts) over the Internet. Each host on the Internet possesses a unique IP address, serving as its distinctive identifier. The primary advantages of subnetting include addressing the IP address shortage on the Internet and, consequently, facilitating the establishment of a fast, efficient, and robust network.

Subnetting involves partitioning a network address space, leading to improved efficiency in address allocation. This process is closely tied to IP addresses, subnet masks, and the utilization of Classless Inter-Domain Routing (CIDR) notation, which is employed for identifying Network Prefixes and Mask [16]. In CIDR notation, the subnet mask is represented by a number indicating the count of ones in the mask, as exemplified by "172.16.2.0/24." Subnetting facilitates communication between connected devices, while routers play a crucial role in enabling communication among different subnets. The size of a subnet is determined by factors such as connectivity requirements and the specific network technology employed.

IP addresses consist of two components: the Network Prefix, also referred to as the Network ID, and the Host ID. The distinction between these two fields depends on whether the assigned network address class falls under Class A, B, or C [17].

Network subnetting is basically two types of subnets, namely Fixed Length Subnetting, also referred to as Classful Subnetting, and Variable Length Subnetting, known as Classless Subnetting. Uniform subnet sizes and an equal number of hosts across all subnetworks characterize length Subnetting. On the other hand, variable-length subnetting exhibits varying sizes, subnets, and numbers of hosts across different subnetworks [17]

The five main subnet classes are Class A, Class B, Class C, Class D, and Class E. Class A, B, and C are commonly employed by various networks, each dealing with a specific range of IP addresses [18]. Each subnet class is uniquely defined by the allocation of bits in its IP address to both network and hosts, each with a default subnet mask. Identification of the class type is based on the number of bits dedicated to the network, and the first octet in the IP address serves this purpose. For instance, **Class A** ranges from 0 to 126 in the first octet, Class B from 128 to 191, and Class C from 192 to 233. Class D spans from 224 to 239, while Class E falls within the range of 240 to 255. Notably, the value 127 is excluded as it designates a loopback address [18]. The classification details, strengths, and weaknesses of each subnet class are presented in Table 2 for network reference.

	Table 2: Review of the subnet class types used on networks				
Subnet Class Types	Ranges of IP address Value assigned	Subnet Mask value	Strength	Weaknesses	
Class A	0-126	255.0.0.0	Their IP addresses are well-suited for vast networks.	It is related to their potential over-allocation of host addresses for smaller networks, resulting in less efficient utilization of address space.	
Class B	128-191	255.255.0.0	Their IP addresses are more appropriate for smaller networks as they allocate 14 bits for the network, leaving only 18 bits for hosts.	The weakness of Class B subnets is their moderate address space allocation, which may lead to address space inefficiency, especially for smaller networks.	
Class C	192-233	255.255.255.0.	Their IP addresses are normally assigned to a very small-sized network	The weakness of Class C subnets is their limited host address space, making them less suitable for large networks or organizations that need a considerable number of host addresses within a single	

				subnet.
Class D	224-239		IP address is almost exclusively reserved for multicasting applications. Multicasting is a networking method enabling one or more senders to communicate with multiple receivers simultaneously.	class D addresses are not intended for typical networking operations. Subnet potential is absent due to the lack of host bits within the Class D address space.
Class E	240-255	255.255.255.255 can be used as a broadcast address (a network address in which devices connect to a multiple-access communications network)	Class E is often cited as having been created for future use, research, and development. Although these IP addresses are reserved, their actual use has never been developed	The weakness of Class E addresses lies in their reserved and experimental nature, rendering them unsuitable for regular network addressing

Despite the high rate of adoption of networks by several companies around the world due to the digital age for the delivery of services, it has resulted in using several technologies and methods to share data and resources to earn a high-quality and timely delivery. Lots of services, resources, and data need to be transmitted at a time to various destinations, which in turn results in the delay of packet (data) to the assigned destination; that is, denial of service is being experienced by the legitimate user at a particular as the packet flow over the network seems very slow or delayed via various digital data transmission links. Therefore, the research aims to contribute to the existing knowledge of different digital data rate links that can be employed by companies by evaluating the selected digital data link rates in the simulated scenarios based on some chosen statistics to determine the best data link rate in the transmission of data and resources over the network and can also be improved to using higher data link rates as they as well consider the cost implication in obtaining them.

# 4. Research Methodology

Having conducted a comprehensive review of the related works to various methods and techniques adopted to simulate the network designs and the application of digital data transmission systems to evaluate their performances on a network, this section discussed the research method, tools adopted, the selected digital data transmission systems, the subnetting class types for each LAN (office), and the network model design.

# 4.1 Justification of the Research Method Used

This paper adopted a quantitative research methodology as it involves the design and simulation of a network model which is based on analyzing and measuring performances in numbers (in graphical form) via the selected statistical parameters. The quantitative research methodology was chosen due to its easy interpretation of the obtained data (results). Another advantage of using a quantitative method for the research is due to its high level of accuracy [19].

# 4.2 Justification for the Adoption of Riverbed Modeler

A Riverbed modeler academic edition installed on a 64-bit Windows 10 system was used to simulate the designed network architecture. It's a software mainly used to analyze and evaluate wireless networks. [20].

Riverbed Modeler was selected for simulating the proposed network model due to its scalability, provision of highfidelity modeling, and well-detailed analysis of a broad range of wired and wireless networks. It helps evaluate enhancements to standards-based protocols and is effective in testing and demonstrating technology designs in real-life scenarios before production. The modeler was used to create and simulate a network model from the given specification, and the breakdown of the model consists of four different LANs (Local Area Networks)/ logical subnets.

# Step-by-step procedures for creating a network in riverbed modeler

- 1. Open the riverbed modeller software
- 2. Click on the file menu to create a new project and the number of scenarios (3) that want to be carried out is set. The type of network (office) to be modeled was selected. After that LAN was selected as the technology for the network setup. Next was selecting a logical subnet to represent each LAN (Manchester, Sydney, Chicago, and Call Center). After which star network topology was selected due to its reliability (if one cable/device fails, others can still work, and it's less expensive)
- 3. For each LAN, application and user profiles were created. Under the application profile, various applications were selected such as file servers, printers, webservers, switches, and so on. The user profiles are comprised of the number of users (nodes) assigned to using some selected applications and connecting them to communicate through the selected data links.
- 4. After all the network setup had been well configured, the simulation duration (time for the topology to rum) was set as well as the statistical parameters for the analysis of the result. Then the simulation tab under the "DES" menu was clicked and the report was generated. When opened, the statistics were viewed in graphical for analysis. This was created for scenario 1, scenario 2, and scenario 3.

# 4.3 The selected digital data transmission systems

Out of several digital data transmission systems embedded in the Riverbed Modeler, three types were selected, namely T1, T3 OCI LINKS, which are forms of long-distance physical network technology used in telecommunication companies.

T1 link (T1 line) is verified to be an effective communication transmission service as it can be used to transmit and receive both data and voice traffic simultaneously over the network at a speed rate of 1.544Mbps. It's also helpful in delivering higher-speed connections, making it easy for networking companies to offer voice and data services. However, its weaknesses led to the selection of a T3 link (DS3 (Digital Signal Level 3)), which is an ultra-high-speed connection used to transmit data at rates up to 45Mbps. It's more connected than the T1 link, as it is effective for regular voice-grade telephone lines. It is fast enough to transmit full-motion, real-time video, and a big database capacity within a busy network. T3 Link is much stronger when configured to carriage multiple T1 links for voice and data services, thus multiplexing numerous digital channels as each channel delivers a bandwidth rate of 64kbps.

**The OCI (Optical Carrier) link is a digital data transmission rate** used for digital signals for carrying on SONET (Synchronous Optical Networking). The bitstream rate of the digital signal defines the transmission rates. Its connections seem to be more than the other two data rates. OC-1 is a SONET line with transmission speeds of up to 51.84 Mbit/s (payload: 50.112 Mbit/s; overhead 1.728 Mbit/s) using optical fiber.

# 4.4 Sketches of the proposed network model

Figure 1 depicts the analogy of the proposed network comprising four logical subnets (LANs) created in the Riverbed Modeler. It was described using scenarios that the four LANs belong to a particular company whose headquarters was in Manchester, UK, and have recently opened regional offices in Chicago, USA, and Sydney, Australia. These regional offices serve as the main service providers in all the reports describing sales made monthly and annually, which are being handled by the headquarters in Manchester, UK. Furthermore, it was decided to perform a simulation in case they will expand with an additional call center. Each LAN consists of several nodes (hosts), network devices such as switches, gateway routers, applications, and profiles used by each user (node). All three offices (Chicago, Sydney, and Call Center) are connected to the Headquarters in Manchester, UK.

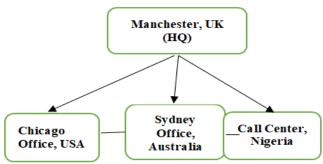


Figure1: The proposed network model sketch

Table 3 describes the breakdown of the model, which was based on scenarios created using four (4) different logical subnets (LANs), as shown in Figure 1. In the Original setup of the company, the Chicago Office (USA) and Sydney Office (Australia) were the main service providers, while Manchester, UK, served as the Headquarters. it was later expanded to create the Call Center Office, which was situated in Nigeria to evaluate the network performance. Several nodes (Users/ workstations) assigned to each logical subnet were shown, as well as various applications (network devices/ services) used by each of them and different User Profiles assigned to different users within the Four logical subnets, which are based on the types of services they offer. From this point of view, the Manchester (HQ) network is assigned 10 hosts because they are responsible for assessing, gathering, and processing every report from both Chicago and Sydney. It is not the main service provider. The headquarters need not have numerous hosts (users) due to all three other offices being connected directly to them for report processing, hence avoiding management overheads on the network.

The four LANs	Manchester, UK	Chicago Office	Sydney Office	Call Center Office
(Subnets ) of the	(HeadQuarters)	(Department 1)	(Department 2)	(Department 3)
Network model				
Number of Users	10	35	25	15
(workstations)				
Selected	10baseT switch, 5 local	100baseT switch, Local	10baseT switch, local	100baseT switch,
Applications used	printers, 1 Web server, and 1	printer, and Email	printer, and Email	Local printer, Email
	FTP server were connected	(heavy) were connected	(heavy) were connected	(light). We were
	to the gateway router using	to the gateway via a	to the gateway via the	connected to the
	serial T1 link	serial T1 link.	T1 line.	gateway via the T1
		The printer and database		line.
		(heavy) serve as local	Printer (light) and	The printer (light)
		while web browsing	database (heavy) serve	and database (light)
		(light-Http) and FTP	as local while web	were selected to serve
		(heavy) are supported	browsing (heavy-Http)	as local, and others
		remotely in Manchester	and FTP (light) are	like web browsing
		(destination	supported remotely in	(heavy) and FTP
		preference).	Manchester.	(light) are <b>supported</b>

Table 3.	The breakdown	of the propos	ed network model
radie 5.	THE DICAKUOWI	of the propos	cu network mouer

User Profiles	Manchester_Profile1: (Text	8 out of 35 users in the Chicago office utilize video conferencing Chicago_Profile1:	8 out of 25 users utilize online voice and video conferencing Sydney_Profile1	remotely.InManchester(destinationpreference)All users run onlinevoice and videoconferencingapplicationsapplicationsSydneyOffice(destinationpreference)Call center_profile1:
	Manchester_Profile1: (Text processing) (This is assigned user1 to user5) Manchester_Profile2: (Text processing + web streaming) (this profile is assigned to user 6 to user10 )	Chicago_Profile1: (Printing and Email (heavy)). This profile is assigned user5, user7, user10, user12, user16, user17, user31, user32, user34, & user35 Chicago_Profile2: (printing (color), database (heavy),( web browsing (light-http), Ftp (heavy) >accessed remotely from Manchester)). This profile is assigned to user8, user9, user11, user13, user 14, user18, user 19, user 21, user22, user23, user24, user25, user26, user27, user28,user29 & user 30 Chicago_Video: (Video conferencing + Chicago_Profile1) This profile is assigned to only 8 users which are user15, user20 & user33.	Sydney_Profile1 (Printing and Email(heavy)) This profile is assigned to user1 to user7 Sydney_Profile2 (printing (light), database (heavy), (web browsing (heavy-http), Ftp (Light)-> accessed remotely from Manchester)). This profile is assigned to user8 to user17. Sydney_Video: (Video conferencing, online voice + Sydney_Profile1). This profile is assigned to user 18 to user25	<pre>(printing, Email (light)) This profile is assigned to user1, user2, user3, user4 &amp; user5.</pre> Call center profile2 : (printing(B/W), database (light), (web browsing (heavy- HTTP), FTP (light) - >accessed remotely from Manchester)) This profile is assigned from user6 to user15. Call center Video:(Online Voice and Video conferencing)

#### 4.5 Subnetting Schemes for each designed LAN

#### 4.5.1 Subnetting for Manchester LAN

Subnet Class A private address was used for Manchester LAN (logical subnet), which consists of only ten hosts because Class A occupies only the First Octet in the IP block. The IP address is 15.10.0.0/8, while the default subnet mask is 255.0.0.0. The calculated subnet results suitable for the Manchester LAN were: Network Address = 15.10.0.192; Usable host addresses range from 15.10.0.193 to 15.10.0.222 (these were assigned to both users in Manchester as well as the selected applications) while the Broadcast Address = 15.10.0.223

# 4.5.2 Subnetting for Chicago and Call Center (Nigeria) LANs

Chicago has 35 hosts, and the Call center has 15 hosts with other additional nodes such as the printers and gateway router (35+15)=50 hosts. IP address is 172.25.31.0/16, and it shows that the Class occupies the first two octets of the network IP block, which means that the default subnet mask for Class B private address space is 255.255.0.0. The calculated subnet results suitable for the Manchester

LAN were: The Network address = 172.25.31.128, the Usable host address ranges from 172.25.31.129 to 172.25.31.190 which is the total of 62 valid hosts IP addresses (sufficient enough for 50 hosts), The Broadcast address is 172.25.31.191.

### 4.5.3 Subnetting for Sydney LAN

Sydney Office consists of 25 hosts (users), and Class C Private Address space was used due to its required number of nodes. The IP address is 192.128.64.0/24 because Class C occupies the first three octets of the IP block. The default subnet mask is 255.255.255.0. **The calculated subnet results suitable for the Manchester LAN were:** The Network Address = 192.128.64.192. The Usable Host Addresses range from 192.128.64.193 to 192.128.64.222 (it can accommodate a reasonable number of hosts needed), while the broadcast Address = 192.128.64.223.

**4.6 Scenarios designed based on the network performance** Three different scenarios were created, starting from the default network architecture, where all three offices (Chicago in the USA, Sydney in Australia, and Situated Call Center in Nigeria) are connected to the Manchester Office, which is the

Company's Headquarters. Figure 2 shows the network architecture of how three offices got connected to Manchester (HQ) via T1 Link using the gateway router. Each LAN is created using a subnet, and all the application and user profiles are configured, showing lists of applications used and assigned to each user through the configured profiles. In Figure 2, the connection was only established between (Sydney- Call Center), (Manchester –Call Center) and (Manchester - Chicago). This formed the first scenario (default scenario) that was conducted.

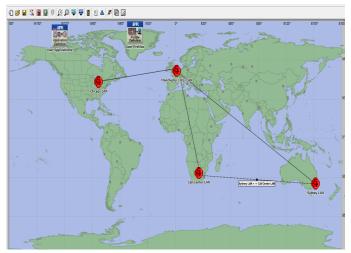


Figure 2: The default network architecture

For scenario 2 and scenario 3, the full connection between all the offices was established and configured. These involve the connections between (Sydney- Call Center), (Manchester – Call Center) (Manchester - Chicago), and (Chicago – Call Center). Figure 3 shows the full network topology where T3 Link was tested in scenario 2 and OC1 Link was used for scenario 3 to evaluate their performances. The scenarios were conducted, various parameters (digital data rate links) were selected for each scenario (Use case), and some statistics were used to effectively carry out the analysis of the network performance.

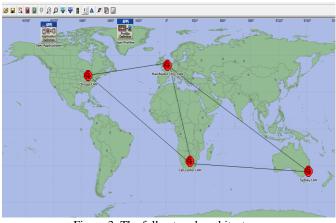


Figure 3: The full network architecture

Table 4 shows that Serial T1 link (DSO) was used for the first scenario in connecting all four LANs for the point-topoint utilization of network services as well as the selected statistics which were used for the analysis of the obtained results across all three Scenarios and these include: Average FTP download response time (it is defined as the elapsed time taken to request and to receive a response); HTTP page response time (it is defined as the average level at which it takes per seconds to load HTTP web page with all its contents); Behaviour of voice and video applications (It is defined as the average time taken for voice packet End to End delay) and Ethernet delay (it is defined as the time taken for all voice and video packets to be received by all nodes on the network).

Scenarios performed on the designed Network Architecture (Model)	Default Scenario (users in Call Centre communicating with the Sydney office)	Scenario 2 (Use case 1) (users in Call Centre communicating with Sydney office + Chicago Office)	Scenario 3 (Use case 2) (users in Call Centre communicating with Sydney office + Chicago Office)
Parameter selected on the riverbed modeler software	The four(4) subnets, Serial T1 link (DSO), point-to-point utilization	The four(4) subnets, Serial T3 link (44.736Mbps), point-to-point utilization	The four(4) subnets, OCI link, point-to- point utilization
Selected statistics	Utilization of T1 Link (64Kbps); Average FTP download response time; HTTP page response time; Behaviour of voice and video applications and Ethernet delay	Utilization of T3 Link; Average FTP download response time; HTTP page response time; Behaviour of voice and video applications and Ethernet delay	Utilization of OCI Link (51.84Mbps); Average FTP download response time; HTTP page response time; Behaviour of voice and video applications and Ethernet delay

# 4.7 Topology setup for the four LANs in the Riverbed environment

Figure 4 shows the overall topology of each LAN, namely the Manchester (HQ) office, Chicago, Sydney, and Call

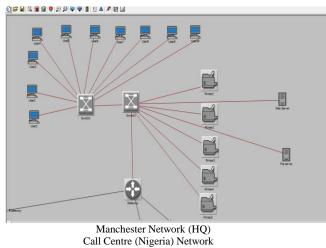
Center offices. It also depicts how they are connected using the assigned digital data rates and several network devices such as printers, Gateway routers, switches, and so on.

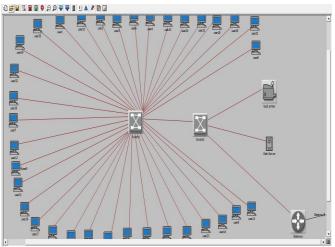
Manchester LAN was set up having 10 users that used more printing applications since one of the primary work assigned to Manchester being the headquarters was to collate and gather reports of sales from all the company's branches. Two switches were used just to serve as a backup in case of failure of one switch all connected using 10BaseT (10Mbps) for network transmission. Two servers are configured for web browsing and FTP which other offices access remotely all connected using a gateway router via T1link (1.544Mbps)

The Chicago LAN consists of 35 users assigned to application profiles all connected using 100baseT(100Mbps) of two switches to avoid network failure, having a local printer that offers printing service to all users and communicates with Manchester LAN using Gateway via T1 link. 8 users accessed FTP server and web browsing (HTTP light) applications from Manchester

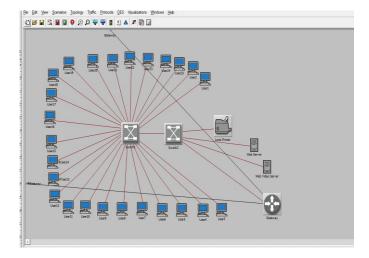
Sydney LAN has 25 users all configured with applications and profiles using 10baseT (10Mbps), then connected to two switches to avoid network failure. It has a local printer, a web server that supports services to the users, and a web video server (a dedicated server) that supports online video conferencing and online voice applications to communicate with the Call Center office (destination preference). Originally, the Call Center office communicated only with the Sydney office via online voice and video conferencing.

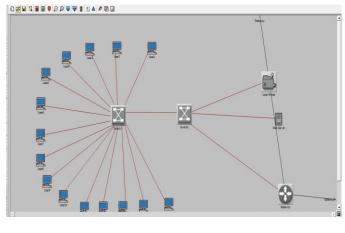
Finally, in the expansion of the company, a Call Center office was created having 15 configured users, a web server for accessing email, and a database connected to two switches to avoid network failure. They communicate with the Sydney office via online voice and video conferencing and also access FTP and web browsing applications remotely from the Manchester office. They are all connected using a 100BaseT (100Mbps) data rate.



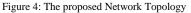


Chicago Network





Sydney Network



# 5. Results and Discussion

Table 5 explains the analysis of the results obtained from the investigation of the network performance using three

different digital data transmission systems (T1 link, T3 link, and OC 1 link) based on the six (6) selected statistics, namely: Utilization of data rate Links (T1/T3/OC1 links),

Average FTP download response time, HTTP page response time, Behaviour of video applications, Behaviour of voice applications and Average Ethernet delay. The analysis of the results was performed based on three Scenarios: the default scenario (Scenario 1), scenario 2, and Scenario 3

# 5.1 Analysis of Scenario 1 (default scenario) results

In the default Scenario (scenario 1) as shown in Table 5 and in Figure 2 which depicts how the connection of the three offices (Sydney, Chicago, and Call Center) linked to the Manchester Office (Headquarter), the T1 link data rate was used in the connection and transmission of data within users in the Call Centre (Nigeria) communicating with the Sydney office (Australia) in which the simulation was executed (run) at the simulated time of 20 minutes.

As shown in Figure 5 (a), based on the utilization parameters (Serial T1 link) for the point-to-point end (from both ends) from Sydney to Call center offices, the utilization of bandwidth rises from 50% to 88% while the utilization of bandwidth from call center to Sydney office decreases from 50% to 25% due to the presence of network clogs. This means that using the T1 link, there is no optimal bandwidth utilization. The results for the average FTP response (figure 5 (b)) show 11 seconds (elapsed time taken to request and to receive a response) before clogs occur and rise to 19 seconds. It can be deduced that the network performance can still be improved to increase the rate of network scalability.

In analyzing the Behaviour of using video application, figure 5(c) shows that within the simulated period, the average in video conferencing packet end-to-end delay was 100 seconds (time taken to send video application packets to the destination nodes), the average in video conferencing traffic sent per seconds increases to 148 packets and the average in video conferencing traffic received per seconds increases to 120 packets. This is to show that due to the level of delay in the traffic flow, clogs in the network are also high from both ends of the offices communicating. This indicates that all users in call center offices communicating with the Sydney office via video conferencing experienced a delay, and the network performance needs to be improved through the use of a higher data rate. The serial T1 link (DSO) data rate was initially used (64Kbps), which is viewed to be pretty low for the transmission of network packets. Using the T1 link, the average Ethernet delay is 85 seconds. This is the time taken for all packets to be received by all nodes on the network. This indicates a high level of delay in the network, as shown in Figure 5(d), and finally, the average level at which it takes per second to load an HTTP web page with all its contents decreases from 1.2 seconds to 0.2 seconds. Figure 5 (e) shows that clogs are present during network communication among users streaming videos.

# 5.2 Analysis of Scenario 2 Results

As shown in Figure 3 and Table 5, scenario 2 (use case 1) involves connecting the Chicago office to communicate with the Call center office which is in addition to the initial communication of the Sydney office only, and the data link rate was changed to T3 link to view the improvement of the

network performance. The average point-to-point utilization from Sydney to the call center increases to 59% while the average point-to-point utilization from the Call Center to Sydney increases to 85%. This means that the network performance communication between the users of Sydney and the call office has improved. Also, the average point-topoint utilization for Chicago to the call center is 88% of bandwidth usage while the average point-to-point utilization for the call to Chicago increases to 59% of bandwidth usage. The average FTP download response time takes 17 seconds and then increases to 35 seconds when network clogs occur. As a result of this, the HTTP page response time also extends from 0.6 seconds to 1.1 seconds and this shows that there is still some level of network delay.

The behavior of the video application during scenario 2, revealed that the average video conferencing packets End to End delay (time taken to send packets to destination nodes) was 92 seconds, the average video conferencing traffic received was 125 packets per second while the average video conferencing traffic sent was 152 packets per seconds. Also, the average voice End-to-end delay was 80 seconds, the average voice traffic received was 1300 packets per second and the average Ethernet delay time was 32 seconds. This also indicates an improvement in the network performance.

# 5.3 Analysis of Scenario 3 results

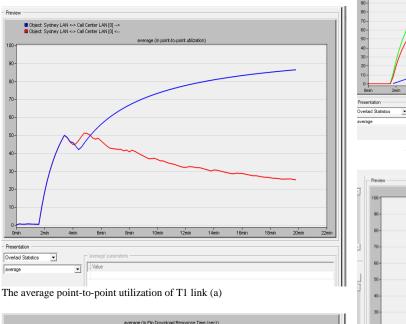
Based on already connected network topology in Figure 3 and Table 5, this third scenario utilized OC1 Link to determine the average point-to-point utilization from Sydney to the Call Center which increased to 88% while the average in point-to-point utilization from the Call Center to Sydney increased to 73%. Also, the average point-to-point utilization of communication from Chicago to the Call Center increased to 73% while the average point-to-point utilization of communication from the Call Center to the Chicago office increased to 88%.

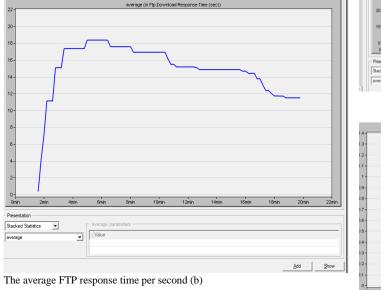
The average FTP response time was initially 0.2 seconds (elapsed time taken between sending a request and receiving the response) but the presence of network clogs raised it to 3 seconds, therefore, the average HTTP page response time increased from 0.6 seconds to 1.5 seconds.

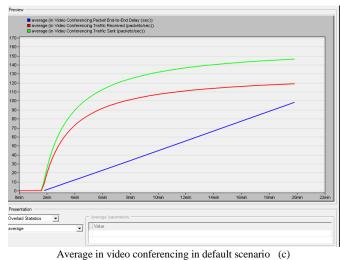
The behavior of video applications shows that the average video conferencing End-to-end delay was 90 seconds, the average video conferencing traffic received was 130 packets per second, and the average video conferencing traffic sent was 155 packets per second. This indicates that the OC1 link brought better network performance

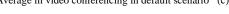
The behavior of voice applications revealed that the average in voice packets end-to-end delay (time taken to send packets) was 92 seconds, the average in voice traffic received was 1250 packets per second, while the average in voice traffic sent was 1500 packets per second. This indicates that the OC1 link for video application causes an end-to-end delay to increase while there was a reduction in the delay of that video conferencing application

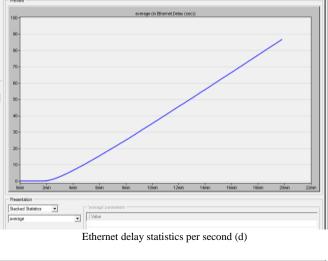
Finally, the average Ethernet delay seconds (this is the time taken for all packets to be received by all nodes was 58 seconds. This shows that there was an increase in delay in the taken to send and receive packets by all nodes as some level of network clogs was still present.

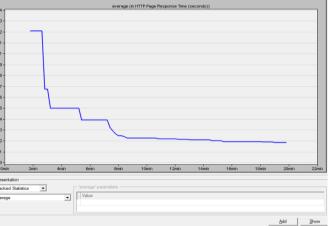












Average HTTP page response time(e)

Figure 5: Results obtained based on the selected statistics

	Table 5: Results of the network performance based on the scenarios				
-	Scenario 1 (default Scenario)	Scenario 2 (Use case 1)	Scenario 3(Use case 2)		
on the designed	(users in Call Centre (Nigeria)	(users in Call Centre	(users in Call Centre		
Network Architecture	communicating with the Sydney	communicating with Sydney	communicating with Sydney		
(Model)	office)	office + Chicago Office)	office + Chicago Office)		
<b>Results obtained using</b>	1. Utilization of serial T1 Link:	1. Utilization of serial T3 Link:	1. Utilization of OC1 Link		
the selected statistics	communication from the Sydney	The serial T3 link data rate is	(51.84Mbps): OCI data rate is		
	office to the Call Center office	higher than T1 link.	higher than T3 link		

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(Nigeria) shows that the usage of bandwidth increased from 50% to 88% while communication from the Call center to the Sydney office decreased from 50% to 25% due to the network clogs

Average FTP download 2. response time: it takes 11 seconds before clogs occur and increased to 19 seconds

3. HTTP page response time decreases from 1.2 seconds to 0.2 seconds. There is a presence of clogs during network communication among users streaming videos

4. **Behaviour** of video applications: packet delay variation increased to 400 bytes within 4 minutes and there is a high level of clogs in the network which increased throughput during the simulated period. The result indicates that all users in call center offices communicating with the Sydney office via video conferencing experienced delay, and the network performance needs to be improved through the use of a higher data rate.

5. Behaviour of voice applications: the average in voice packet End-to-end delay is 100 seconds (time taken to send packets to destination nodes). The packets increased from (1200 to 1500) packets/seconds.

6. Average Ethernet delay: it takes 85 seconds for all packets to be received by all nodes on the network. This indicates a high level of delay in the network.

#### **Overall summary:**

There's a high rate of traffic delay, and the data rate (T1 link) needs to be improved.

Also, using the T1 link, bandwidth is not optimally utilized.

utilization from Sydney to the call center increases to 59% while the average in point-topoint utilization from the call center to Sydney increases to 85%.

\*The average point-to-point utilization for Chicago to the call center is 88% of bandwidth usage while the average pointto-point utilization for the call center to Chicago increases to 59% of bandwidth usage.

\* Though using the T3 link may be costly to the company, hence, it helps to reduce the level of network clogs (delay).

2. Average FTP download response time: the average FTP response time is 17 secs (elapsed time taken between sending a request and receiving the response) before network clogs occur and increases to 35 seconds. This indicates that there's an increase in the FTP response time.

3. HTTP page response time: The average HTTP page response time increased from 0.6 seconds to 1.1 seconds, and there was a high rate of clogs in the network. Behaviour of video 4.

applications: the average in video conferencing packets End to End delay was 92 seconds. (time taken to send packets to destination nodes). The average in video conferencing traffic received was 125 packets per second and the average in video conferencing traffic sent was 152 packets per second.

\* Comparing this to when using the T1 link, the delay of packets has been reduced and the number of packets per second received has increased.

5. Behaviour of voice applications: the average voice End-to-end delay was 80 seconds, the average in voice traffic received was 1300 packets per second and the average in voice traffic sent were 1500 packets per second.

\* It has indicated here, that by using the T3 link, the time taken for delay of packets is reduced and the number of traffic received has increased.

6. Average Ethernet delay: The Ethernet delay average was 32secs, and this is the time taken

\*The average in point-to-point \* The average in point-to-point utilization from Sydney to the Call center increases to 88% and the average in point-to-point utilization from the Call center to Sydney increases to 73%.

> \* The average in point-to-point utilization of communication from Chicago to the Call center increased to 73% usage of bandwidth, while the average in point to point utilization of communication from the Call center to the Chicago office increased to 88%

2. Average FTP download response time: the average FTP response time was 0.2 seconds (elapsed time taken between sending a request and receiving the response) before network clogs occurred and increased to 3 seconds.

3. HTTP page response time: The average HTTP page response time increased from 0.6 seconds to 1.5 seconds, and there is a high rate of clogs in the network.

4. Behaviour of video applications: the average video conferencing End to End delay was 90 seconds, the average video conferencing traffic received was 130 packets per second, while the average video conferencing traffic sent was 155 packets per second

Behaviour of 5 voice applications: the average in voice packets end-to-end delay (time taken to send packets) was 92 seconds, the average in voice traffic received was 1250 packets per second, while the average in voice traffic sent was 1500 packets per second.

\* Using the OC1 link for video application causes an end-to-end delay to increase while there's a reduction in the delay of that video conferencing application

6. Average Ethernet delay: The average Ethernet delay was 58 secs; this is the time taken for all packets to be received by all nodes. This shows that there's an increase in delay in the taken to send and receive packets by all nodes

#### **Overall summary**:

\* it has clearly shown that Using the OC1 link helps to reduce the end-to-end delay for the users of video conferencing than the T3 data rate; that is, they can receive

for all packets to be received by all nodes <b>Overall summary</b> : * It has indicated that using the T3 link for network communication of the company will help improve the network scalability. * T3 link helps to reduce the level of clogs when using video conferencing applications and voice applications and end-to-end	their packets at a lesser time. * However, there are still issues arising from using the OC1 link, such as the increase in end-to-end delay when using online voice applications, and the HTTP page response time also increased over the network.
delay while the HTTP page response time increases.	
* Though there are still clogs, not as much as when using the T1 link. Therefore, another higher	
link. Therefore, another higher data rate was used to have more improvements.	

# 6. Conclusion and Recommendation

Having run simulations of three different Scenarios and analyzed the results, it has been revealed that using the Serial T1 link for communication among all the company's offices did not optimize the network performance, and there were clogs (delays) specifically for users using video and voice applications to communicate (Chicago, Call Center, and Sydney). The average video conferencing packet end-to-end delay was 100 seconds (time taken to send video application packets to the destination nodes), the average video conferencing traffic sent per second increased to 148 packets and the average in video conferencing traffic received per second increased to 120 packets. This indicates that the users experienced delays in the timely use of video and voice applications.

Therefore, the T3 Link, which has a higher capacity for transmitting packets over the network was used in place of the T1 link. Chicago was also connected to communicate with the call center office via video-conferencing application. Viewing the results, it was revealed that using the T3 link helps improve the network performance. The results reveal that the average video conferencing packets End to End delay (time taken to send packets to destination nodes) was reduced to 92 seconds, the average video conferencing traffic received was raised to 125 packets per second while the average video conferencing traffic sent increased to 152 packets per seconds. Also, the average voice End-to-end delay was 80 seconds, the average voice traffic received increased to 1300 packets per second and the average voice traffic sent was raised to 1500 packets per second. In addition, the average point-to-point utilization was up to the optimal level, although there are still clogs in the network, and it was lower than when the T1 link was used.

Given results obtained using T1 and T3 Links, the OC1 link was introduced, which proved to have a higher capacity for transmitting network packets. The results obtained from its simulation revealed that the average point-to-point utilization of bandwidth during network communication between Sydney and to Call Center became better as it increased to 88% and 73% respectively. Also, the average video conferencing End-to-end delay was reduced to 90 seconds, the average video conferencing traffic received increased to 130 packets per second, and the average video conferencing traffic sent was raised to 155 packets per second. This indicates that the OC1 link brought better network performance. However, there are still issues regarding voice applications, in which the delay time of sending and receiving packets increased.

Finally, comparing all the scenarios conducted has shown that for future work, more improvements can still be made to the network architecture of the company if a higher data rate can be used as well as increasing the number of statistical parameters to obtain better results. Although it will result in higher expenditure, the network communication among the offices of the company will be improved.

#### **Data Availability**

All details about the subnet breakdown for the overall network model are available on request

#### **Conflict of Interest**

The authors declare that there is no conflict of interest with anyone for publication of this work.

# **Funding Source**

None

# **Authors' Contributions**

Author-1 Conducted the design and simulation of the network model and other experimental work, as well as comprehensively explained the generated results and concluded the work.

Author-2 explored the introduction and existing scholarly works and revised the manuscripts at the final stage.

#### Acknowledgments

We offer our sincere thanks to the divine guidance of God Almighty, which illuminated our path and inspired our work. Furthermore, we extend our gratitude to the International Journal of Scientific Research in Computer Science and Engineering for their invaluable feedback and constructive suggestions, which have significantly contributed to the clarity and visibility of our research efforts.

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