Distributed Multimedia Applications in Quality of Service for Wireless Wide Area Network

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Abstract

QoS is a primary consideration in data transmission over wireless networks. Recent studies have shown great advances in the field of QoS research due mainly to the emergence of multimedia networking and computing. The incessant requirement of organisations across the globe for voice and video conferencing using client/server model communication over an IP network calls for adequate research and analysis in to the organisational network model with a view to understanding what best serve the organisation before implementing it. This study uses OPNET modeller to design and simulate two cloud based networks. The two scenarios in which scenario 1 runs traditional applications and scenario 2 runs multimedia applications were compared. The results of the two scenarios were benchmarked with similar output results from literature on the performance of QoS parameters (Throughput, Delay and Loss) that contribute towards the development of an efficient QoS network. The work finding shows that multimedia applications (voice and video) result to very high throughput and are sensitive to delay which results to data loss whereas Traditional applications(email, file transfer, web browsing) can use minimum throughput and with the average data Loss and are normally insensitive to changes in delay.

Keywords: QoS, multimedia applications, traditional applications, wireless network, Opnet.

1.0 Introduction

Quality of service (QoS) plays a vital role in multimedia applications (video and audio). This is because in Multimedia data transmission, QoS must be guaranteed from network layer up to the end system. Developing a suitable Distributed multimedia system that can deliver QoS support and end to end time is subject to many challenges that include; inexact delays in the communication layer, system storage, communication resources and the processing and memory resources. Variability in data rates and sensitivity to losses due to data transmission to a range of locations are common characteristics of distributed multimedia applications [1]. Recent studies in the field of Multimedia networking [2] have shown significant interest in supporting connection-level QoS for multimedia applications in wireless networks. There has been various distributed call-admission control schemes proposed to adjust the desired bandwidth so that a low cell-overload probability is maintained. The demand for providing multimedia applications with a QoS guarantee in wireless networks is becoming more difficult and challenging due to inadequate bandwidth resources and user's mobility. Wireless networks offer a more sophisticated communication with a lower bandwidth and high error rate and Latency[2]. The incessant requirement of organisations across the globe for voice and video conferencing using client/server model communication over an IP network calls for adequate research and analysis in to the organisational network model with a view to understanding what best serve the organisation before implementing it.

Optimum Network known as Opnet is used in this study because it enables easy means of developing models from real world network, and it supports all major network types and technologies that allow you to design and test various scenarios with reasonable output results. Opnet modeller is an object oriented simulation tool that was created in 1987[3]. It offers a visualised simulation environment for networked environment and it has been used in analysing new protocols and applications.

The new wireless LAN model will operate on client/server network model that will be divided into two scenarios and all the colleges are represented as subnets. The three subnets will be connected through an IP32_cloud internet. Cloud computing is a model that stores information permanently on servers over the internets and temporarily on cached in client like computers and other devices [4]. IP network can be made up of subnets that are inter-connected to one another with different network (IP) address [5]. The first scenario will run applications like email, HTTP, FTP, file print and web browsing while scenario 2 will run similar applications with scenario 1 but with additional voice and video applications. The performance of the two scenarios will be compared and the overall results of the two scenarios which are:Throughput, Delay and Data dropped will be used to evaluate and compare the discussed output with my results.

2.0 Quality of Service Overview

QoS has no specific definition; its definition depends on the exact perspective where the word has been used [6]. In the context of this reseach, Quality of service can be defined as "a set of quantitative and qualitative characteristics of a network necessary to achieve the required functionality of applications and to satisfy the user" [7]. It may also be defined as a collection of service requirements that allow you to control bandwidth for network traffic thereby guaranteeing adequate service level for data transmission. In other words, QoS is a set of capabilities that a network must meet to guarantee adequate service level for data transmission. Moreover, QoS allows the specification of the required parameters to control traffic and improve network performance over a wireless network [8].

To deploy QoS into a network, there are certain parameters that are necessary to control the amount of traffic sending over the network. These parameters include: bandwidth, delay, jitter, loss, throughput and error rate [9].

2.1 Quality of Service Parameters

QoS parameters provide the ability to control the amount of traffic, priority, reliability and speed over a network[10], the common QoS parameters that come to mind when deploying QoS in a network are discussed below:

- BANDWIDTH

Bandwidth measures the rate of traffic in a network. It is expressed either in bits per second or in hertz. Bandwidth in bits per second determines the rate at which a channel, a link or network can transmit in bits per second while Bandwidth in hertz determines the range of frequencies a channel can pass [11].

- THROUGHPUT

Throughput determines the speed of data/packet sent through a network. Throughput and Bandwidth seems to be confusing, but they are entirely different. While Bandwidth determines a potential rate of a link, Throughput determines the speed of the data/packet [11].

- LATENCY

Latency determines the time delay of transmission and reception of data/packet from source to destination [12].

- JITTER

Jitter is described as the variation in the latency or time delay between packet deliveries as a result of congestion and queuing along the network path.

- LOSS

Packet loss is described as a loss of data when traversing the network. Loss can happen due to network congestion or errors during transmission. [12]

In this study, three QoS parameters have been choosing, namely: Bandwidth, delay and loss to evaluate the performance of multimedia and traditional applications in wireless networks. However, different applications require different set of service requirements. For example, multimedia applications require minimum throughput and maximum latency in a network.

2.2 Applications and QoS Requirements

It is important to discuss the applications upon which the imposition of QoS is required. They are simply categorised in to traditional and multimedia applications. Traditional applications refer to applications that have more stringent requirements to Loss while multimedia applications are those that have fewer requirements on Loss. Both traditional and multimedia applications have a stringency of their network requirements.

The table below,[13], provide a summary description of the applications and their parameter’s requirements.

Table 1.0: Applications & their parameter requirements[13]

Application	Bandwidth	Delay	Jitter	Loss
Email	Low	Low	Low	Medium
File sharing	High	Low	Low	Medium
Web access	Medium	Medium	Low	Medium
Remote login	Low	Medium	Medium	Medium

Audio on demand	Low	Low	High	Low
Video on demand	High	Low	High	Low
Telephony	Low	High	High	Low
Videoconferencing	High	High	High	Low

From table 1.0 above, the following can be inferred:

1. Bandwidth: Applications that requires high bandwidth are file sharing, video on demand and video conferencing while email, remote login, audio on demand and telephony applications do not require much bandwidth.
2. Delay: Unlike email and file sharing, telephony and video conferencing are very sensitive to delay.
3. Jitter: Audio and video applications are very sensitive to jitter. This is because of variation in transmission time (Latency). If there is a delay of 1 or 2 seconds, before a packet data reaches its destination, the result will not be clearly seen and audible. Applications such as email, file transfer and web access are not sensitive to jitter.
4. Loss: Unlike multimedia applications (audio and video), traditional applications are sensitive to loss because they need to be delivered correctly. However, congestion and packet loss cannot be prevented in a situation whereby the network has inadequate bandwidth and too much delay. Loss and Jitter can be restored by retransmission of data and buffering packets at the receiver respectively [13].

2.3 Quality of Service in Wireless Networks

Wireless network is any form of computer network characterised by the absence of cables in its connectivity. It is a system in which two or more equipment locations are connected avoiding the use of cables thereby reducing the overall cost of connectivity and communication by getting rid of most of the physical infrastructure and its associated labour cost [14].

Wireless Networks are broadly used for a wide range of purposes, with many applications of varying QoS constraints making use of wireless networks[15]. Example of applications with QoS constraints are voice over IP, video streaming etc. These Applications have common requirements of QoS parameters on Throughput, Delay, and Delivery ratio.

2.4 QoS for Distributed Multimedia Applications

Quality of service (QoS) plays a vital role in multimedia applications (video and audio).This is because in Multimedia data transmission, QoS must be guaranteed from network layer up to the end system. Challenges in developing a suitable Distributed multimedia system that can deliver QoS support and end to end time are due to ambiguous delays in the communication layer, system storage, communication resources and the processing and memory resources. Variability in data rates and sensitivity to losses due to data transmission to a range of locations are common characteristics of distributed multimedia applications [1]. For distributed multimedia applications to deliver end-to-end QoS, a good processor resource is needed because multimedia interaction might results to poor QoS and extreme exploitation. Certainly, quality degradation can occur while a multimedia session is in progress due to network saturation or host congestion. Therefore increasing the utilisation of the system processor resources is important to conform to variations in the resources or load.

QoS allows description of quantitative parameters (e.g. Jitter, Delay, bandwidth, Loss) and the qualitative parameters that are useful in the estimation of level of service by the user. However, there are various QoS layers (Network, OS and Devices) that describe the actual end-to-end level of service.

To regulate the system resources, user QoS parameters are interpreted into application level parameters and then set into system-level parameters (Network, processor). The QoS mapping is done by the resources management components of the framework, which allows the user to identify the QoS requirement. Moreover, QoS mapping does not allow mapping QoS parameters into underlying layers parameters. The QoS parameters are described as sets (name, value). [1].

2.5 Distributed Multimedia Applications in Wireless Networks

"In wireless network, Multimedia data transmission inherits also all the characteristics and constraints related to the propagation to the free space" [16]. As best-effort services are good at Datagram traffic [17], there has been tremendous and unique request for wireless networks to be able to support elastic and inelastic traffic so as to guarantee a good and satisfactory QoS in wireless networks. The demand of QoS provisioning problem in wireless network is as a result of inadequate bandwidth and host mobility. Instances of such problems in wireless networks is that if a mobile host is placed into the network that can guarantee/satisfy all its demands, it might however move to a new cell that has inadequate or no resources at all to satisfy its requirements/needs.

3.0 METHODS

3.1Description of the Wireless Network Model

The new wireless LAN model will operate on client/server network model that will be divided into two scenarios and all the scenarios will consists of three subnets represented as Katsina, daura and dutsinma subnets. The first scenario will run applications like email, HTTP, FTP, file print and web browsing while scenario 2 will run similar applications with scenario 1 but with additional voice and video applications. As a result of voice and video applications in scenario 2, a multimedia server will be added in each subnet of scenario 2 to support voice and video applications.

Furthermore, scenario 1 will consist of wireless workstations, wireless server and a printer and access point that will connect with a switch then the switch will be connected to a router for internet access. Likewise in scenario 2, it will consist of wireless workstations, wireless server then a printer, access point and multimedia server that will connect to a switch and the switch will be connected to router for internet access. Application configuration and profile configuration are set at global level so that all the subnets will access their services.

This section describes the implementation of the wireless wide area network (WWAN) scenario by using OPNET simulation tool. The general format of the wireless WAN consists of two scenarios in which scenario 1 will run non-real time applications while scenario 2 will run both real time and non-real time applications.

SCENARIO 1: The format of scenario 1 consists of the following three subnets:

1. KATSINA SUBNET: As the main subnet,it consists of WLAN subnets of 3 sub subnets as follows:

Each subnet is represented by 10 wireless workstations, 1 wireless server, a printer and access point that are connected to a switch. All the subnets are connected to a switch via 100BaseT_link and a router that is connected to the internet via PPP_DS 1 link. The representation of each campus scenario is implemented in OPNET as shown below:

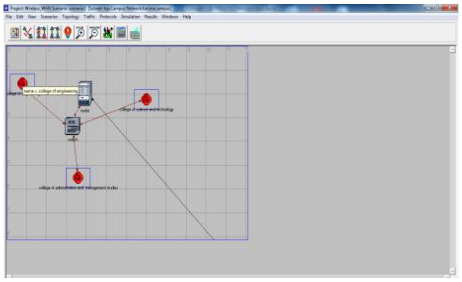


Figure 1: showing the three subnets

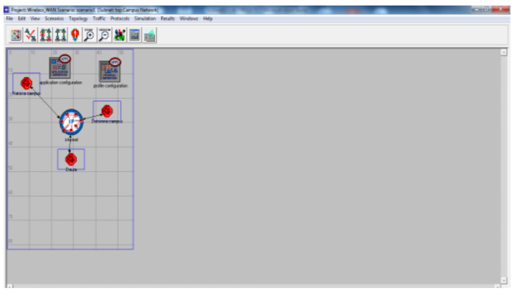
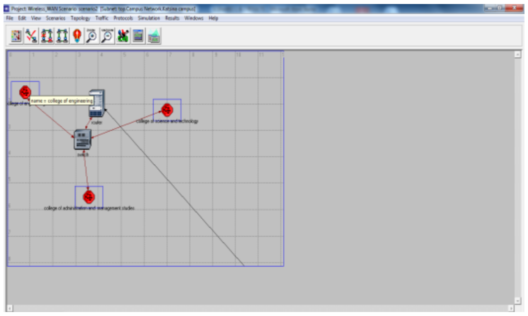


Figure 2: one of the subnets for scenario 1

2. DUTSINMA AND 3. DAURA SUBNET: These consist of 10 wireless workstations, a wireless server, router, access point and a printer. The router, printer and access points are connected to a switch via 100Base_Tlink and router is also connected to the internet via PPP_DS 1 link.

SCENARIO 2: This scenario runs the same application as scenario 1 but with additional voice and video applications. Therefore, a multimedia server is added in each subnet to support video and voice applications.



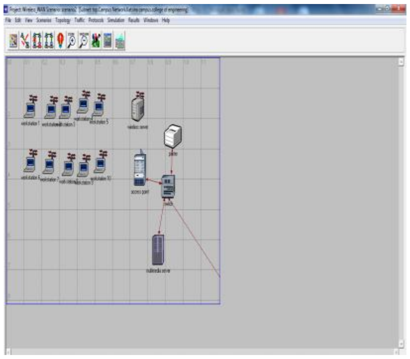


Figure 3: one of scenario 2 subnet

Figure 4:Scenario 2 subnets connected to switch then switch to router

3.2 Basic Components for the Implementation of the Scenarios

The basic components that are necessary for the above mentioned networks are listed and described in the table below:

Table 2.0: Basic Components for the Implementation of the Scenarios

COMPONENT	MODEL	DESCRIPTION
Application configuration	App. Config	This node is used to specify different tier names used by the network model and the specified application name will be used to while creating user profile on the profile configuration
Profile configuration	Profile.config	This node is used to create user profile. It specifies the applications used by a particular group of user
Wireless	Wlan_wkstn_adv	Represent a workstation with client_server

workstations		applications running over TCP/IP and UDP/IP
Access point	Wlan_ethernet_router_adv	Is a wireless LAN based router with one ethernet interface.
Subnet	Subnet	Allows you to display a network through abstraction.
Wireless server	Wlan_server_adv(fix)	Represents a server node with server applications running over TCP/IP and UDP/IP station server applications directly using the services offered by AAL
Internet	IP32_cloud	Represents an IP cloud supporting up to 32 serial line interfaces at a selectable data rates through which an IP traffic can be modelled
Switch	Ethernet16_switch_adv	Represents a switch supporting up to 16 ethernet interfaces. The number of connections is limited to 16
Link	100BaseT_adv	Represents an Ethernet connection operating at 100 mbps
Link	PPP_DS1_int	Connects two nodes running IP. Data rates is 1.544 mbps
Printer	Ethernet_printer_adv	Represents a server node with server applications running over TCP/IP and UDP/IP.

4.0 Results and Discusion of Results

4.1 Introduction

This is the overall results of the scenarios and their discussion. The performance of the two scenarios were compared, after choosing the metrics, the simulation was done for 300 seconds and then the results were gathered. This is the overall results of the two scenarios discussed above, which are:

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 2 Issue 10, October - 2013

1. Throughput 2. Delay and3. Data dropped. From the following graphs, the red line represent scenario 1, while the blue line represent scenario2. Scenario 1 consist of non-real time applications namely: File transfer(FTP), Email,File print, Database and Web browsing(HTTP) whereas Scenario 2 consist of real time applications that includes voice, video and non-real-time applications of scenario 1.

4.2 Throughput

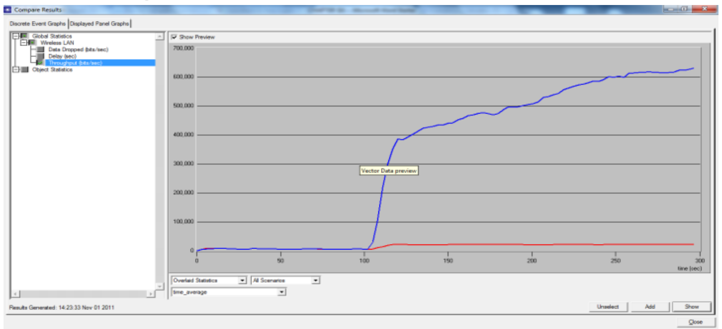


Figure 5.0: Throughput result

In fig. 5.0 above, The Y-axis represents the throughput which is the number of bits against X-axis that represent simulation time in seconds. It is observed from the figure above that at 0 seconds, no data was carried, at 0-100 bits per seconds; the amount of data carried was the same. After 100 seconds, the overall Throughput of scenario 2 increases to 390 00bits, which is equivalent to 47.6kbits. At 120 seconds, the throughput of Scenario 2 rises to 602 00bits which is equivalent to 587.89kbits. On the contrary, the overall throughput of scenario 1 is stabilised throughout of the transmission.

4.3 Delay

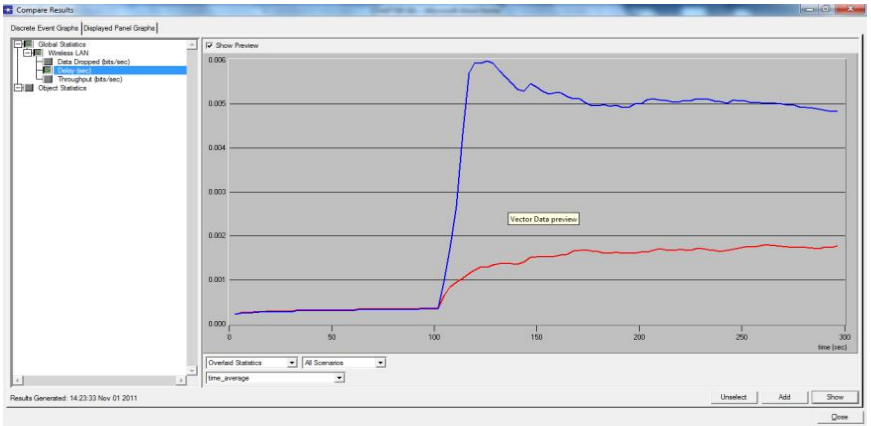


Figure 6: Delay Result

The figure 6 above shows that the Y-axis represents the Delay time in seconds against X-axis that represent simulation time in seconds. At 0seconds, no data was carried out and from 0-100 seconds the amount of data carried was the same for all scenarios. After 100 seconds, the overall Delay of scenario 2 suffers a sudden increase to 6 *ms (*millisecond*)* whereas scenario 1 rises to 1.4 *ms* and then stabilise. At 140 seconds, the amount of data carried in scenario 2 was delayed by 5.2 *ms*. Thus, it can be concluded that the overall delay of scenario 1 is minimum compared to scenario 2.

4.2 Data dropped

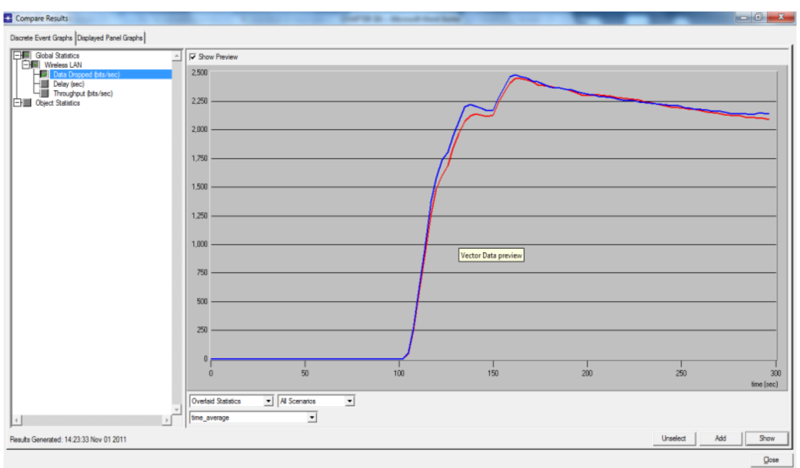


Figure 7: Data Dropped result

In fig. 7 above, The Y-axis represents the Data dropped which is the number of bits against X-axis that represent simulation time in seconds. It is observed from the figure 4.2 above that at 0 seconds, no data was carried, at 0-100 bits per seconds; the amount of data carried was the same for both scenarios. At 110 seconds, the overall data dropped of scenario 1 and 2 increases to 2150= 2.09kbits and 2200= 2.14kbits respectively. At 150seconds, data dropped decreases to 2125=2.07kbits for scenario 1and 2150=2.09kbits for scenario 2. At 170seconds, the data dropped rises to 2450=2.39kbits for scenario 1 and to 2500=2.49kbits for scenario2. It can be concluded that the average data dropped of both scenarios increase and decreases at the same time though scenario 2 is bit higher than scenario1.

Results from throughput

- SCENARIO 1: The overall throughput of scenario 1 as shown in figure 4.0.1 is steady throughout of the transmission. This is because traditional applications such as file transfer email, web browsing that are run in this scenario does not require high throughput and congestion is not likely to occur as stated in chapter two and four by Stallings and Tanenbaum.
- SCENARIO 2: The overall throughput of scenario 2 is high and varies with time; this is because scenario 2 consists of multimedia applications that increase the load of the network as seen in figures above, Throughput increases as load increase. Though it is stated by Stallings in chapter two that multimedia applications require minimum steady throughput in data transmission, in this case the Throughput is high due to presence of Traditional and Multimedia applications.

Results from delay

- SCENARIO 1: the overall delay of scenario 1 is low , it can be concluded that the overall delay of scenario 1 is minimum compared to scenario 2 because the applications run in scenario 1 requires low delay as well as they are not quite sensitive to delay as stated by Tanenbaum and Stallings in Table 1.0.
- SCENARIO 2: the overall delay of scenario 2 is high due to presence of multimedia applications that are quite sensitive to delay as specified by Stallings. It can be concluded that the overall delay of scenario 2 is high compared to scenario 1.

Results from data dropped (Loss)

In chapter two of this work, Stallings stated that real time applications which are voice and video are quite sensitive to Loss whereas Tanenbaum in Table 2.1 indicated that traditional applications require medium Loss. From this study, the average data dropped of both scenarios increase and decreases at the same time though scenario 2 is bit higher than scenario1.

Basically, based on these simulation results, the following findings were inferred:

- Multimedia applications i.e. voice and video results to a very high throughput and are sensitive to delay which results to data loss (see figure 5, 6 and 7
- Traditional applications such as email, file transfer, web browsing etc, can use minimum throughput with average data Loss and are normally insensitive to changes in delay as seen in figures above.
- Throughput: as seen in Figure4.0, at a minimal load, throughput is steady. There is a direct relation between load and throughput such that an increase in load is accompanied by a corresponding increase in throughput. The increase in throughput seen in scenario 2 is due to the presence of multimedia and traditional applications that might have caused congestion in the network.
- Delay: From the graph one can observe that the delay is high in the real time application than in the traditional application; this is because for real time application it's obvious that all the packets have to reach the destination within few seconds of delay, therefore the delay is always high. Traditional applications tolerate minimum delay as Shown in Table 1.0. From figure 4.1, at a small load, the amount of delay is small. As the load increases, the delay also increases due to presence of multimedia and traditional applications in scenario 2 .
- Loss: for the data lost, you can observe that there is a slight difference in data lost/drop; i.e. the data drop is high in the real time applications than in the traditional application. This is what has been expected even though the difference is not much as shown in Table 1.0.

5.0 Conclusions

The foregoing study discussed QoS for Wireless Networks, QoS for Distributed Multimedia Applications, Wireless Networks and IEEE, Challenges and Limitations of QoS in wireless Networks. It also investigated on performance of Throughput, Delay; Data dropped (Loss) and IP cloud in a network. It also designed and simulated network scenarios in Opnet and investigated the simulation results on QoS primary parameters (Throughput, Delay and Data dropped (Loss)).

After designing the network and obtaining the simulation results, the following findings were inferred:

- Multimedia applications i.e. voice and video results to a very high throughput and are sensitive to delay which results to data loss see figure 5, 6 and 7.
- Traditional applications such as email, file transfer, web browsing etc, can use minimum throughput with average data Loss and are normally insensitive to changes in delay as seen in figure 5, 6 and 7 .
- Throughput: as seen in Figure 5.0, at a minimal load, throughput is steady. There is a direct relation between load and throughput such that an increase in load is accompanied by a corresponding increase in throughput. The increase in throughput seen in scenario 2 is due to the presence of multimedia and traditional applications that might have caused congestion in the network.
- Delay: From the graph one can observe that the delay is high in the real time application than in the traditional application; this is because for real time application it's obvious that all the packets have to reach the destination within few seconds of delay, therefore the delay is always high. Traditional applications tolerate minimum delay as Shown in Table 1.0. From figure 4.1, at a small load, the amount of delay is small (scenario1). As the load increases, the delay also increases (scenario 2) as shown in Figure 6 and Figure 7.
- Loss: for the data lost, you can observe that there is a slight difference in data lost/drop; i.e. the data drop is high in the real time applications than in the traditional application. This is what has been expected even though the difference is not much as shown in Table 1.0.This shows that multimedia applications are more sensitive to loss than traditional applications because in traditional application no matter the duration of delay, the packet will be delivered correctly, unlike in multimedia applications in which if there is delay of some seconds before a packet reaches its destination the result will not be clearly seen and heard.

This implies that it is of great importance to simulate a network before it's been set up in order to identify the requirements of the applications that the network will run and to detect any problem that might arise in real life.

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