Optimization of concurrent VOIP calls across Wireless Mesh Networks

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By

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Declaration

I, Olusola Olorunnisola, declare that this thesis research study entitled ‘Optimization of concurrent VOIP calls across Wireless Mesh Networks’ is my work. It is being submitted in partial fulfilment of the requirements for the Master of Computing Degree: Information Technology at the Tshwane University of Technology, Pretoria. I further declare that all sources cited or quoted are indicated and acknowledged using a comprehensive list of references.

Signature Date

…….. ……..

Dedication

I dedicate this study to my family, especially my mother, who gave me support and who encouraged me in my moments of weakness to keep pushing to where I am today.

Acknowledgment

First and foremost, I want to thank God for the strength he has given me to carry out this research. Second, I would like to express my thanks to my supervisors, Dr. T.E. Mathonsi, and Dr. D.P. du Plessis, for their constant direction, warm encouragement, critical comments, patience, and support. Their guidance and suggestions were always kind and constructive. I will also not forget the support received from my friends, Mr. J. Kasiroori, and Mr. Blessing. Finally, I would like to express my gratitude to my entire family for their encouragement and motivation.

Publications

Parts of this research study have been submitted and accepted for publication in the following journal and conference proceedings:

Conference Proceedings

1. Olusola Olorunnisola, Topside E. Mathonsi and Deon P. Du Plessis, “Performance Optimization of Concurrent VOIP Calls Across Wireless Mesh Networks”. Proceedings of the 2022 International Conference on Computational Science and Computational Intelligence (CSCI'22), 14 – 16 December 2022, Las Vegas, USA, Page 1130 – 1137, ISBN-13: 979-8-3503-2028-2, DOI 10.1109/CSCI58124.2022. Available at: https://american-cse.org/csci2022-ieee/csci2022.zip or https://ieeexplore.ieee.org/xpl/conhome/1803739/all-proceedings.

Journal Article

1. Olusola Olorunnisola, Topside E. Mathonsi and Deon P. Du Plessis, “An Algorithm to Optimize Concurrent VOIP Calls Across Wireless Mesh Networks”. Submitted to the Journal of Advances in Information Technology.

Abstract

Voice over IP (VoIP) is an emerging technology with huge potential. The need for wider coverage which we are accustomed to in circuit-switching telephone communications i.e., Global System for Mobile communication (GSM) and Code-Division Multiple Access (CDMA) has brought about transmission of the Voice over Wireless Mesh Networks (VoWMN). The VoWMN implementation faces many problems due to limited bandwidth availability, jitter, and interference (both internal and external). Even so, it has some benefits such as the ease of deployment, expansion, wider coverage, and reduction in cost over voice communication. In this study, the adaptive VOIP optimization algorithm was designed, implemented, and evaluated based on the following parameters, namely, packet loss, network throughput, jitter, and latency. The proposed algorithm was compared with three existing VoIP schemes: 1) end-to-end aggregated traffic, 2) ROHC compressed-only traffic, and 3) plain traffic without aggregation or compression. The various performance measurements in the experiments were captured using the *seventh.cc* data-capture module which was introduced in Network Simulator - 3 (NS-3). Data analysis was achieved using Python, and the metrics were then plotted using Gnuplot.

The proposed algorithm performed better than all other algorithms in the following test categories: jitter, packet loss, and throughput. However, the algorithm had the lowest performance in latency, with the highest latency at 31.59; whilst the scheme with the lowest had 18.43, a difference of 13.16mbps. Packet loss was lower than the compared algorithms. The algorithm had only 1.4% packet loss and jitter of 0.62435ms which was also the lowest among the schemes. The adaptive VOIP optimization algorithm produced a better network throughput than the other schemes. In this study, the mean opinion score (MOS) was calculated. The MOS measures the quality of the call using packet loss rate, latency, the R-factor, and jitter. The proposed algorithm scored 4.103 which is very good VoIP quality. Thus, we believe that we have achieved the main objective of this study; hence, the work will find applications in industries, society, and academia.

Keywords: Adaptive VOIP optimization algorithm, *Compression, Aggregation, MOS, GSM, ROHC, VoWMN.*

Glossary

This section is an alphabetical listing of contextual definitions of key terms used in the study.

- **Broadband internet** – the transmission of wide bandwidth data over a high-speed internet connection.
- **Concurrent VOIP calls** – the number of simultaneous calls that a VOIP solution can handle.
- **Internet protocol (IP)** – a set of rules for routing and addressing data packets so that they can travel across networks and reach the correct destination.
- **Multi-path Wireless Network** – a routing technique simultaneously using multiple alternative paths through a network in order to boost fault tolerance, increase bandwidth, and enhance security.
- **Packet switch network (PSN)** – a computer communications network that groups and sends data in the form of small packets.
- **Public service telephone network (PSTN)** – the collection of interconnected voice-oriented public telephone networks. The network brings together networks that are operated by national, regional, or local telephone operators.
- **Real-time communication (RTC)** – a near-simultaneous exchange of information over various telecommunications services from the sender to the receiver.
- **Session initiation protocol (SIP)** – a signalling protocol used for initiating, maintaining, and terminating real-time voice, video, and messaging sessions.
- **Voice communication** – (a) communication by word of mouth; (b) the communication of sound over a distance using wire or wireless telephones and related technology
- **Voice over Internet Protocol (VoIP)** – the transmission of voice and multimedia content using an internet connection. VoIP enables users to make voice calls from a digital device.
- **Wireless Mesh Networks (WMN)** – communications networks made up of radio nodes organized in a mesh topology.

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CHAPTER ONE: INTRODUCTION

1.1 Background

Humanity has depended on voice communication from time immemorial. Voice communication has been regarded as the best method of real-time communication. The public service telephone network (PSTN) was developed to enhance voice communication because of its importance to man. Technological advancement later gave rise to transmitting of voice using internet protocol (IP) technology over a packet switch network, termed voice over IP (VoIP) (Okech *et al.,* 2012). The Federal Communications Commission (2021) defines VoIP as “a technology that allows you to make voice calls using a broadband internet connection instead of a regular (or analog) phone line”. Figure 1.1 presents a diagrammatic illustration of the equipment required to make a VoIP call. VoIP is mostly preferred over other alternatives such as the public switched telephone networks (PSTNs) because of its cost-effectiveness and efficiency (Montazerolghaem, 2021).

Figure 1.1: Equipment required to make a VOIP call (Montazerolghaem, 2021)

A VoIP call between two participants requires three basic types of protocol: signaling, media transmission, and media control. The Federal Communications Commission (2021) points out that VoIPs vary, with some services only allowing those who use the same service to make calls; others allow calling any telephone number whether local, long-distance, mobile, or international. Some VoIP services only work over a computer or a special VoIP phone connected to a computer with an internet connection. Other services are compatible with the traditional phone that is connected to a VoIP adapter (Federal Communications Commission, 2021).

The signaling protocols, e.g., H.323 protocols, and the session initiation protocol (SIP) establish, maintain, and terminate a connection, which should be understood as an association between applications. However, media transmission and media control are handled by protocols called real-time protocols which move voice data between the participants. IEE 802.11 defines wireless mesh networks (WMN) as one of the emerging types of network technology that is readily available. Where delivery of voice communication is difficult to deploy, wireless mesh networks make deployment easy and reduce costs (Okech *et al*., 2012). Access points (AP) communicate with one another via wireless links which make up the multi-path wireless network in WMN; large areas can thereby be covered with wireless access at low cost. IEE 802.11 comprises many standards that can be deployed as WMN standards, e.g., 802.11b, g, n, and s. This research focuses specifically on the use of 802.11ac due to its bandwidth capacity, 5GHz frequency, and throughput of up to 7GB (Khalil *et al.,* 2020).

A routing protocol employed when deploying a voice-over wireless mesh network (VoWMN) is important, as the VoIP traffic moves within the multi-hop path of the WMN. An effective routing protocol will certainly enhance VoWMN performance (Andrreev *et al.,* 2010). The VoWMN is burdened with many drawbacks. According to Olariu *et al.* (2016), voice codec, jitter, delay, and packet loss affect the quality of the VoIP call or the number of VoIP calls across the WMN. Research has been conducted in the area of Voice over WMN. Much research was done on the implementation of packet aggregation and little on combinations of various aggregation and compression techniques. None made use of the WMN with a high-performance standard such as 802.11ac.

Despite the noted efficiency and cost-effectiveness of VoIP, room for improvement could not be ruled out. Therefore, this research sought to develop simulations meant to improve the performance of Voice over WMN in an 802.11ac scenario. Such can be achieved by looking for a way to increase the number of concurrences while eliminating or decreasing the impact of jitter, delay, and packet drop.

1.2 Motivation

Voice is an important aspect of communication. The earliest way of communicating long distances was through the plain old telephone system (PSTN), making use of a technology known as circuit switching. Voice over IP address (VoIP) technology came later and replaced the PSTN which required a dedicated resource (Okech *et al.,* 2012). VoIP requires implementation by a more robust network infrastructure. Due to this requirement, a network infrastructure technology that will improve the coverage and performance of VoIP was developed, known as WMN (Matus *et al*., 2017).

The VoIP brought about a cheaper option than the pricey PSTN. VoIP's wide coverage and improved performance of VoWMN was a relief to already burdened indigent communities around the world. This research will demonstrate that the implementation of packet aggregation, compression, and improved packet routing will improve VoWMN performance.

1.3 Research Problem

VoWMN is a prime candidate for solving many issues of VoIP transmission enumerated by other researchers. Optimal use of available bandwidth in WMN which is not yet achievable is an impediment to improving the concurrent call across WMN. Researchers Zulu *et al*. (2012) suggest that implementing packet aggregation with iterative cycles increases the number of concurrent VoIP calls across WMN when transmitting VoIP. However, packet loss is still noticeable, and packet size is still not being controlled. In addition, Meeran *et al.* (2017) make use of the various IEE 802.11 standards (e.g., b, g, n, s) in the deployment of VoWMN. However, due to low capacity, the number of calls decrease compared with the high bandwidth throughput of IEE standards such as 802.11ac.

Therefore, to eliminate or at least decrease the impact of the jitter, delay, and packet loss to a minimum, a new algorithm was proposed. This algorithm makes use of the packet aggregation method; compression with a good routing protocol, and with a high

capacity WMN standard such as 802.11ac was proposed by this research to ensure optimal performance of the VoWMN.

1.4 Research Questions

In order to address the problem statement and purpose of the study presented earlier, this research seeks to address the main research question, which reads as follows:

How can a packet aggregation and compression algorithm to improve the performance of VoIP calls across WMN be developed?

The main research question is subdivided into the following sub-questions:

a) What is the past and current state of the VoWMN?
b) How can packet aggregation and compression algorithms be designed to improve the performance of VoIP calls across WMN?
c) How can the proposed algorithm be implemented in order to increase the concurrent call across WMN?
d) How can the proposed algorithm be validated against existing algorithms in the WMN?

1.5 Research Aim and Objectives

Then the main aim of the study was:

To develop a packet aggregation and compression algorithm which improves the performance of VoIP calls across WMN.

In order to achieve the main objective of the study stated above, the researcher set out the following research sub-objectives:

a) To review the past and current state of VoWMN.
b) To design a packet aggregation and compression algorithm which improves the performance of VoIP calls across WMN.

c) To implement the proposed algorithm to increase the concurrent calls across WMN.
d) To validate the proposed algorithm against existing algorithms in WMN.

1.6 Scope of the Study

The main focus of the study was to improve the bandwidth utilization and throughput which in turn increases the number of concurrent VoIP calls in WMN without impacting the call quality.

1.7 Significance of the Study

Creswell (2014) describes the significance of a study as its importance to different groups. This study is important to various stakeholders as described below.

1.7.1 Telecommunications experts

The study proffers a just-in-time varied solution to some problems affecting the transmission of VoWMN. It is believed that this recommendation will be useful to telecommunications experts as for quite some time they have been grappling with minimizing jitter and delay.

1.7.2 The academia

The research findings and recommendations provide a supplement to the existing body of literature on VoWMN. Besides adding to the existing literature, this study provides a foundation upon which future researchers interested in VoWMN can take over their research, improving on what has been proposed by this study.

1.7.3 Governments and policymakers

It is anticipated that findings from this study will influence policy on VoWMN; the expectation is that the recommendation derived from the simulations would be useful to governments as they develop standards for the telecommunications sector.

1.7.4 Society at large

If adopted, the proposed recommendations from this study would greatly benefit society, as VoIP communication would be greatly improved. The quality and cost of voice communication are likely to be influenced by this study.

1.8 Research Methodology

Whereas research is the “logical and systematic search for new and useful information on a particular topic”, research methodology refers to the “systematic way to solve a problem” (Rajasekar *et al.*, 2013). The researcher outlines the design science research methodology. The methodology and rationale for using it in this study are briefly explained below.

1.8.1 Design science research

The study employed the design science research methodology. Brocke *et al*. (2021) define design science research (DSR) as “a problem-solving paradigm that seeks to enhance human knowledge via the creation of innovative artifacts”. Using this methodology, an innovative, purposeful artefact for VoIP traffic transmission was created in order to address the problem identified. In addition, the designed artefact was evaluated for its utility for the VoIP transmission problem. This was necessary for the researcher to ensure that the purpose of the study had been met. In order to ascertain novelty and contribution to research, the researcher tested the artefact’s ability to solve the identified VoIP problem. This was by providing a more effective solution compared with earlier attempts to solve the same problem.

1.8.1.1 Research methodology flow chart

The flow chart presented in Figure 1.2 below provides a synopsis of the way in which the design science research methodology was employed in the study.

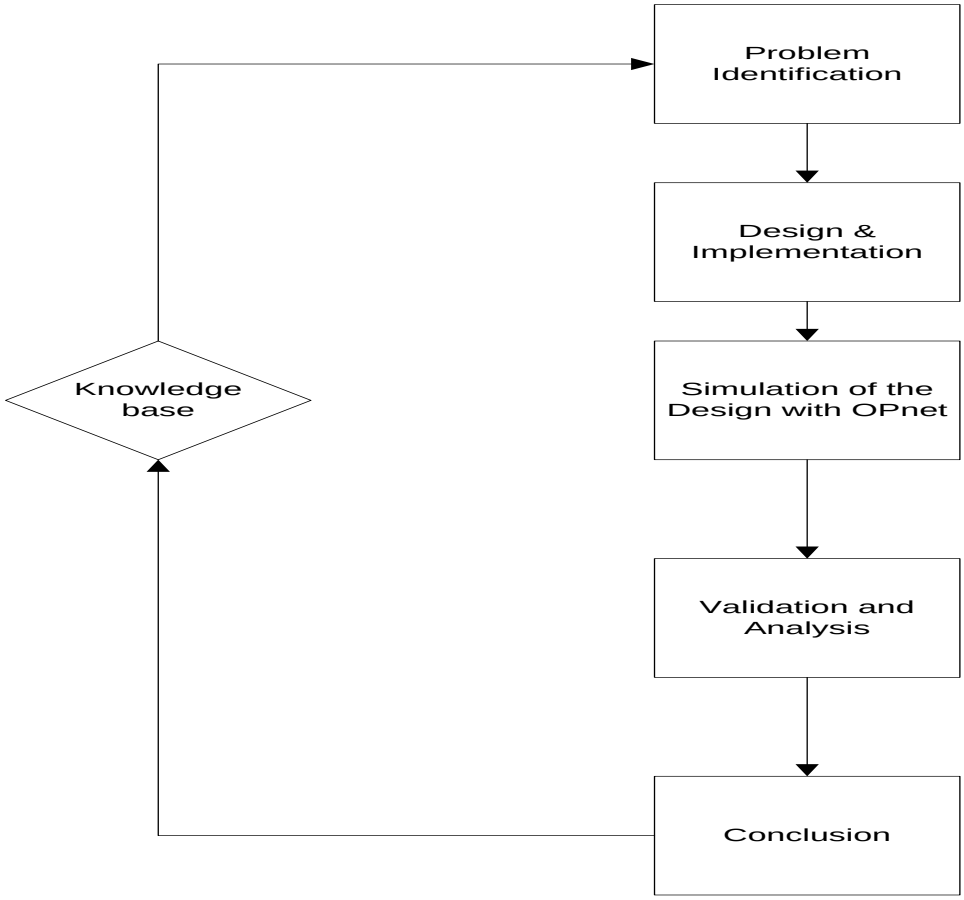

Figure 1.2: Design science flow chart

The flow chart in Figure 1.2 provides an outline of the following:

i) Presentation of the problem: Poor quality VoIP call when transmitting via VoWMN.

ii) Design of VoIP over wireless mesh network with 802.11ac radio e.g., Figure 1.3 below.

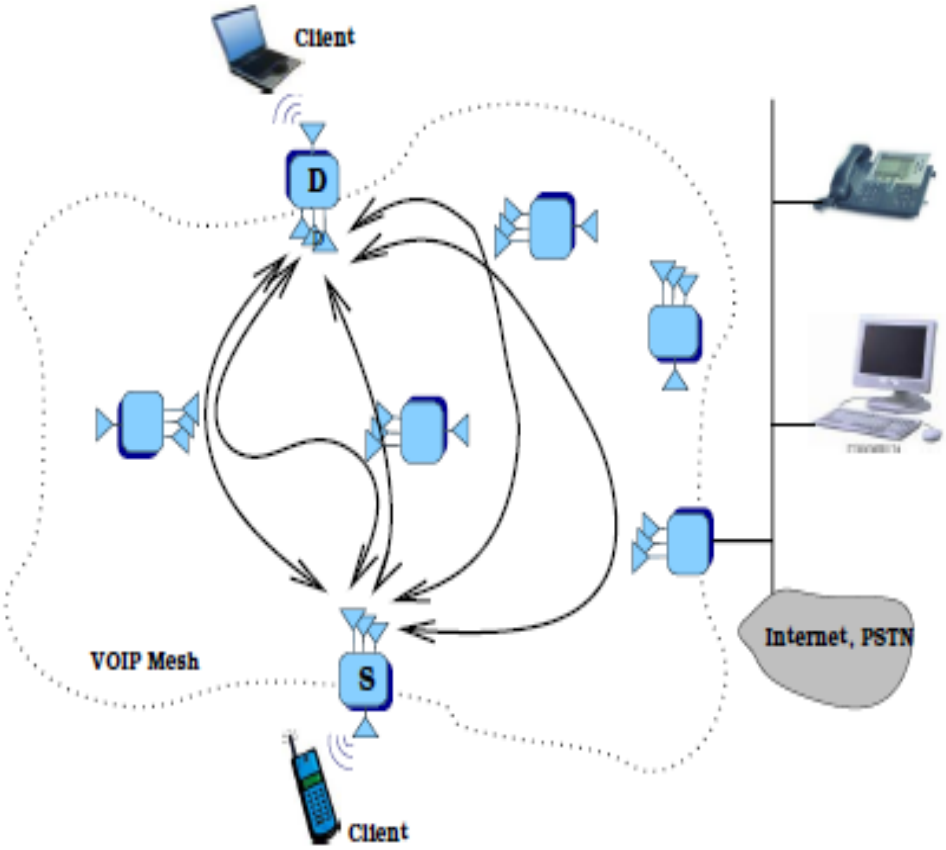

Figure 1.3: VoIP wireless mesh networks (Niculescu *et al*., 2006)

Design development and implementation of VoIP traffic aggregation & compression

a) Aggregation: a collection of small packets into a large packet to enhance the overhead-to-payload ratio. The method that will be employed in this research is the accretion aggregation scheme.

b) Compression: is a standardized method of converting the large overhead to only a few bytes. The compression method that will be employed in this research is robust header compression.

c) The improved routing protocol known as hybrid wireless mesh protocol (HWMP) will be employed.

iii) Model validation, simulation, and analysis of the result.

Simulation of VoIP network with OPNET/NS3 simulator

Implementation of the below network topology in OPNET/NS2 simulator includes the transmission of voice traffic and the application of various factors such as bandwidth constraint, jitter, and interference. Such factors normally affect VoIP traffic transmission over wireless mesh networks, thereby deriving various parametric values. Analysis of the various parametric values against standard and number call values will give rise to the hypothesis that can validate and provide a conclusive answer to the problem of optimization of VoIP traffic across wireless mesh networks. The below flow chart can also be used to illustrate the sequence of activity involved in the successful simulation.

iv) Conclusion and formulation of a knowledge base for future improvement.

The above pointers give a full explanation of what the flow chart in Figure 1.2 entails.

1.9 Organization of the Dissertation

The dissertation is structured as outlined below.

Chapter One: Introduction

Chapter One provides an introduction and background to the study of VoIP & Wireless Mesh Networks. It presents the research problem, objectives of the study, research questions, justification of the study, and an overview of the research methodology adopted for the study.

Chapter Two: Literature Review

Chapter Two provides a literature review of VoIP over wireless mesh networks. The literature is thematically reviewed according to the research objectives of the study. Further, existing literature on the advantages and disadvantages of VoIP over wireless mesh networks and reasons for the bad quality of voice when transmitting over wireless networks is reviewed. Possible ways of alleviating the challenges are presented.

Chapter Three: System Design and Architecture

The chapter describes the research methodology used in the study. It outlines the research approach, research design, algorithms, and techniques used to collect and analyse the data from the simulation of VoIP over wireless mesh networks conducted.

Chapter Four: Implementation

Chapter Four presents the implementation of the proposed algorithm in NS-3.

Chapter Five: Analysis of Results

Chapter Five presents the analysis of the study findings gathered during the research work, namely, the simulation of VoIP over WMN.

Chapter Six: Summary, Conclusions, and Recommendations

The chapter presents a summary of the research findings, conclusions derived from the study, and recommendations. Possible areas for future research for those in the academia, and improvement for those in the VoIP over wireless practice, are also provided.

1.10 Chapter Summary

This chapter introduced the study by providing an outline of the research. The conceptual background of VOIP and challenges encountered with existing VOIP systems were presented. The chapter presented an outline of the research problem and the purpose of the study. For easy comprehension, the purpose of the study was broken down into objectives and research questions. The significance of the study to the telecommunications sector, academia, government, policymakers, and the public in general was presented, along with definitions of key terms used in the study. The study continues in the next chapter in which a review of related literature is outlined.

CHAPTER TWO: LITERATURE REVIEW

2.1 Introduction

A literature review is conducted in order to identify existing knowledge of a particular topic, thereafter identifying gaps upon which new research can be carried out (Serra, 2015). A literature review, therefore, helps the researcher to establish the way in which the current study is related to those conducted in the past (Oliver, 2012). This literature review therefore, set out to ascertain what past studies on VOIP have established. The review then identifies research gaps on which this study is premised. The process of reviewing related literature was also useful in ensuring no repetition of a previous study, a method suggested by Oliver (2012).

This chapter reviews the literature on the optimization of concurrent VOIP calls across the wireless mesh, as reflected in the research topic. More precisely, this literature review will provide an overview of the VOIP concept, giving a basic understanding of the concept; thereafter, the literature review is presented thematically. The themes are derived from the research objectives of the study.

The remainder of this chapter is structured as follows: Section 2.2 presents an overview of VoWMN and VOIP. Section 2.3 provides an outline of the WMN and VoIP routing protocols. Section 2.4 presents an overview of the state of VoWMN. Section 2.5 reviews the aggregation types available. Section 2.6 is a presentation of related work on VoWMN and VOIP, and Section 2.7 presents the chapter summary.

2.2 Overview of VoWMN and VOIP

Voice communication has remained an important mode of communication over the years. Human beings depend on voice communication for socialization; but more importantly, for business interactions. Considering the value of voice communication in human interaction, people have made sure that voice communication is considered significant. Civilization has reached out for technological developments that ensure that voice communication is enabled. Voice over Internet Protocol (VOIP) is one such technological

development that man has invented, continuing to improve on such to ensure that voice communication remains the bedrock of interaction among human beings. Kaur *et al.* (2019) cited Viber and Skype among public services that are using VOIP technologies.

VOIP refers to the technology of transmitting voice over packet-switched networks (PSNs) such as the Internet (Olariu, 2013; Kaur *et al.*, 2019). In simpler terms, VOIP is a technology that allows people to make telephone calls over the Internet (Jalendry *et al.*, 2015; Shaw *et al*., 2016). According to Olariu (2013), the technology is sometimes referred to as internet telephony or internet protocol (IP).

Chong and Matthews (2004) provide a comparison of the traditional telephone and VOIP systems. The authors point out that the traditional telephone system or public switched telephone network (PSTN) provides dedicated end-to-end circuit connections for the duration of each call by caller. Conversely, a VOIP system allows real-time transmission of voice signals in the form of data packets over internet protocol networks (Chong *et al.*, 2004).

A call using VOIP technology is therefore significantly different from that of a traditional telephone system because of the architecture involved, as illustrated in Figure 2.1 below.

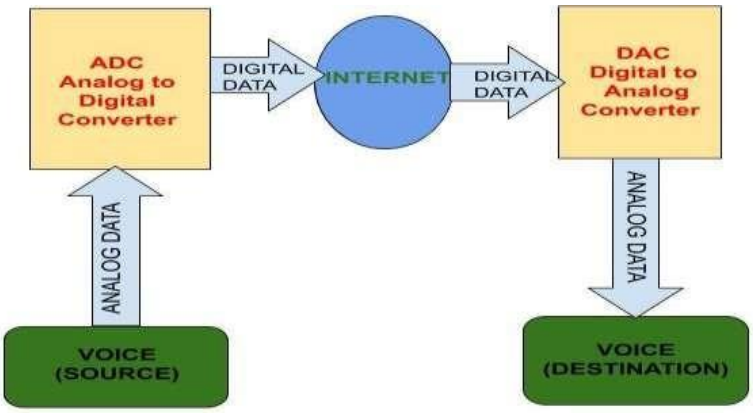

Figure 2.1: Voice transmission in VoIP network

As Chong *et al.* (2004), Jalendry *et al.* (2015), and Abualhaj *et al.* (2016) observed, VoIP has become such a popular technology because of its usefulness in communication. On

one hand, Montazerolghaem (2021) reports that technology has become popular because of the benefits that users derive from it, namely:

i. Best quality of services
ii. Relatively low cost, and
iii. Reliability.

Jalendry *et al.* (2015), Kaur *et al.* (2019), Shahdad *et al.* (2016), and Roy *et al.* (2021) acknowledge that, on the other hand, VOIP technology has its own challenges that include:

i. Network or bandwidth capacity
ii. Network architecture
iii. System design
iv. Performance
v. Reliability
vi. Availability
vii. Scalability
viii. Obsolescence and service life
ix. Security
x. Regulatory issues
xi. Electricity and backup power, and
xii. Quality of service.

Therefore, while the technology proves very useful in human interaction, there are a number of challenges that require attention if more benefits are to be derived from VOIP.

Voice over internet protocol wireless mesh networks (VoWMN) are described by Kim *et al*. (2006) and Okech et *al*. (2012) as a very attractive way of extending the network coverage into the dead zone, in cases where the wired network is not easy to install. As Olwal *et al.* (2008) put it, “the combination of wireless mesh networks (WMN) with VoIP is an attractive solution for enterprise infrastructures”. According to Kim *et al.* (2006), and

Alabady *et al.* (2013), mesh networks have a comparable advantage over wired local area networks (LANs) because of:

i. Ease of deployment
ii. Ease of expansion
iii. Better coverage
iv. Robust against node failure, and
v. Reduced cost of maintenance.

Liu *et al.* (2021) describe WMN as “a multi-hop wireless network made up of communication nodes following a mesh topology”. Also, Carrillo *et al*. (2019), and Akyildiz *et al.* (2005) point out that wireless mesh networks enable individuals to be continuously online, thereby solving problems associated with wired networks.

The VoWMN communication paths are maintained among wireless mesh nodes, with each mesh node consisting of at least two wireless interfaces, one for the clients, the other for backhaul connection (Kim *et al.*, 2006). Figure 2.2 presents a typical VoWMN network architecture.

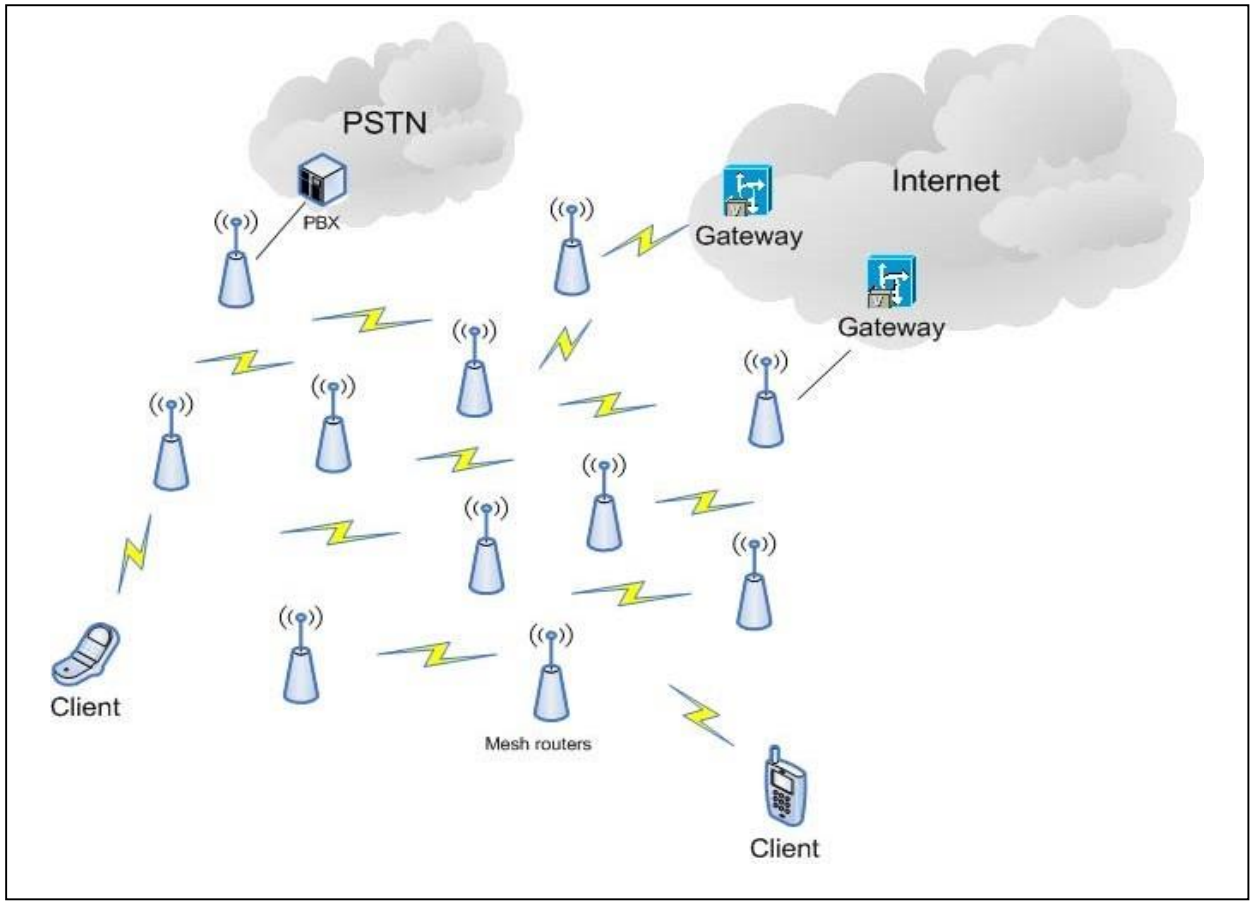

Figure 2.2: VoWMN network architecture

2.3 WMN and VoIP routing protocols

Simulations conducted by Meeran et al. (2014) have established that node mobility can increase packet loss, delay, and jitter. The simulations also revealed that VOIP implementations in wireless mesh networks are more efficient in cases where there is no background traffic. The authors, therefore, recommend that VOIP implementations in wireless mesh networks have mobile nodes. Such nodes, in turn, support background traffic, thereby increasing the quality of service standards and assurance of packet data delivery.

Similarly, studies by Meeran *et al.* (2016) reveal that VoIP quality of service in WMN can be improved by identifying the integration choices and inclusion of supportive mesh nodes. The above-mentioned authors used a network simulator; and experiments were

conducted on three main scenarios with mesh nodes in no-mobility, partial mobility, and full mobility deployment. Findings from this study showed that the proposed approaches would improve VoIP quality in terms of 5-point MOS rating-scale by 0.2 in no mobility, 2.2 in partial mobility, and 0.9 in full mobility scenarios (Meeran *et al.,* 2017).

Tchernykh *et al*. (2019) present a more interesting scenario. These authors propose the use of cloud-based systems as an alternative solution to enabling cost-quality optimization of VOIP systems. Tchernykh *et al.* (2019) developed three configurable parameters for model based on bin packing, namely, utilization threshold, rental threshold, and prediction interval (PI). These mechanisms cope with various objective preferences, workloads, and cloud properties. As the authors propose, the mechanisms can be dynamically adapted to environmental changes.

Furthermore, Chethan *et al.* (2018) proposed an optimized channel allocation (OCA)-based adaptive informant factor (AIF) model that accesses channel information. The OCA-AIF responds whenever interference is detected via AIF. Responding to the detected interferences ensures that packet transmission remains at the optimal level, being closely monitored and maintained.

Simulation is important as it provides a cost-effective and safe way of solving real-life problems. This section provides a review of related literature on selected simulated scenarios involving both aggregation and compression algorithms with the advanced routing protocol. The simulations are summarized as follows:

i. Amiri *et al.* (2020) make reference to the data-aggregation back pressure routing (DABPR) scheme. In the proposed scheme, overlapping routes are instantaneously aggregated for effective data transmission, and to prolong the life of the network. The routing algorithm which has multiple attributes and decision-making metrics includes five elements: cluster-head selection, maximization of event-detection reliability, data aggregation, scheduling, and route selection.
ii. Darabkh *et al.* (2019) proposed a balanced power-aware clustering and routing protocol (BPA-CRP) in which the network topology divides the sensor field into

equal-sized layers and clusters. This enables any cluster to operate a batch without necessarily setting up overhead. The BPA-CRP technique assigns four different broadcast ranges for each sensor.

iii. Fahad *et al.* (2019) prefer a compressed fuzzy logic-based multi-criteria ad hoc on-demand distance vector (AODV). Here routing decisions are dependent on number of relays, distance factor, direction angle, and vehicle-speed variance. An earlier study by Jain *et al.* (2016) presented similar recommendations.

iv. Castellanos *et al.* (2016) advance the QoS-aware routing protocol with adaptive feedback scheme for video streaming in mobile networks. The protocol has mechanisms for detecting link failures in a route in order to re-establish connections so that quality of service is maintained. Alghamdi (2015) is of the opinion that a load-balancing ad hoc on-demand multipath distance vector (LBAOMDV) routing protocol is more appropriate. According to Alghamdi (2015), "LBAOMDV regulates the fair usage of both node energy and available bandwidth by exploiting the availability of multiple paths for data transfer".

v. Bhattacharjee *et al*. (2013) recommend an energy-efficient routing algorithm that strives to strike a balance between data traffic among the nodes and network lifetime by the use of the shortest path tree (SPT) and minimum spanning tree (MST) under the auspices of distributed energy balanced routing (DEBR) and the shortest path aggregation tree (SPAT)-based routing protocol.

2.4 State of VoWMN

Past studies suggest that VoWMN is an evolving phenomenon. The studies reveal that although the technology has been very useful in communication, greater improvements are anticipated so that greater efficiency and effectiveness will be achieved. Shahdad *et al*. (2016) and Meeran *et al.* (2016) admit that mesh network technology is highly dynamic and unpredictable. The authors point out that wireless mesh networks (WMN) belong to the IEEE 802.11s standard, an emerging network system suitable for various applications, including harsh environments such as military fields and emergency relief situations (Meeran *et al.*, 2016).

Soloviev *et al.* (2018) observe that the increased volume of voice traffic in IP-based networks has presented problems with routing voice calls vis-à-vis quality, cost, and security implications. There are concerns that, as the volume of voice traffic escalates, quality is compromised, a guarantee of ideal security decreases, and users of the system may not realise benefits for the costs incurred. However, these factors cannot be overlooked, because of the importance they have on system selection and usage by users.

Abualhaj *et al.* (2016) report that wireless mesh networks (WMN) are increasingly gaining popularity for voice calls, replacing traditional communication systems. The researcher reiterates that VoIP over WMN suffers from inefficient bandwidth use: attaching a 40-byte RTP/UDP/IP header to a small VoIP payload and 841 µs delay overheads of each packet in WMN. The authors then suggest VoIP packet multiplexing as the most appropriate solution to the problem.

Considering their robustness, reliability, and speed, WMN has been adopted to enhance services in the following, according to Shahdad *et al.* (2016):

i. Broadband home network
ii. Enterprise-wide networking
iii. Metropolitan area networks
iv. Building Automation
v. Transport Systems
vi. Health and Medical sciences
vii. Security and surveillance systems
viii. Warehouse systems, and
ix. Disaster reporting and emergency networking.

2.5 Aggregation Types

Zulu (2012) defines packet aggregation as “a means of combining small multiple packets together to form a larger packet”. Various aggregation algorithms have been proposed for the purposes of increasing WMN capacity; one such proposal was by Akyurek *et al.*

(2018). According to Akyurek *et al.* (2018), packet aggregation is a promising technique for increasing WMN efficiency. Zulu (2012) goes further to provide a summary of the two ways in which aggregation can be conducted, namely:

i. Hop-to-hop aggregation, which involves the de-aggregation of packets at one hop, followed by a hop-to-hop aggregation at the next. This implies that after packets are de-aggregated, they are aggregated when ready for transmission.
ii. End-to-end aggregation, in which packets are only aggregated on the sending and receiving nodes. Aggregation only occurs on packets heading for a common destination, while de-aggregation occurs on the receiving node. The other nodes are only responsible for forwarding packets until they reach the final destination, where de-aggregation takes place.

Whilst Marwah *et al.* (2011) are of the view that "packet aggregation is justified up to a certain extent only as after that most of the users get dissatisfied". Akyurek *et al.* (2018) propose an optimal packet aggregation algorithm at three levels of flexibility, namely:

i. Per application data stream
ii. Per packet, and
iii. Under different network conditions.

Akyurek *et al.* (2018) also report that tests of their proposed algorithms yielded a 60 percent increase in energy savings, being 1.55 times higher in information freshness.

2.6 Related Work

Neves *et al.* (2018) have an alternative proposal for increasing WMN capacity. The authors suggest that optimal classification of voice packets is necessary for enhancing the quality of voice communications over priority-enabled networks when poor transmission conditions are encountered. This implies that voice packets have to be assigned as either high or low priority depending on their relevance; such that packets of high priority are given preference over those in the low priority category. According to

Akyurek *et al.* (2018), “the proposed method is based on a dynamic programming optimisation algorithm that finds the optimal subset of *m* high-priority voice segments in each utterance of size $n > m$.”

A study by Olariu *et al.* (2016) proposes what the authors refer to as a “Delay-aware Packet Prioritization Mechanism” (DPPM), whose aim is to “uniformly distribute the Quality of Service (QoS) level across all Voice Over IP (VoIP) calls in a Wireless Mesh Network (WMN)”. The technique prioritizes VoIP packets based on the amount of queueing delay that has been accumulated across multiple hops within the WMN. VoIP packets in the queue for longer are given higher priority over less delayed VoIP packets. The proposed DPPM is also used to ensure that voice-call quality and capacity are enhanced in combination with Wi-Fi frame aggregation.

An earlier study by Azevêdo Filho *et al.* (2012) made similar suggestions. These authors propose a packet aggregation technique known as “holding time aggregation” (HTA) in which the system is highly adaptable to diverse link conditions of wireless settings. The technique “uses an adaptable packet retention time to allow relay nodes to explore aggregation opportunities on a multi-hop path”, thereby keeping jitter and total delay within set application limits.

A recent study by Huang *et al.* (2019) proposed the implementation of a queuing delay utilization for on-path service aggregation. This scheme uses the queuing delay of packets in order to reduce overhead transmission volume and communication. Furthermore, each packet is divided into forwarding packets and aggregating packets, with the forwarding packets being used to complete the service of aggregating packets to minimize the transmission volume and communication overhead without additional latency. Routers handle some requests for the data centre during peak periods, thus reducing the traffic load on the data centre during peak times. Huang *et al.’s* (2019) study finding show that SAQD reduces the data-centre workload by 55.8%–66.26%. Response delays are reduced by 31.33%~51.41%, which leads to a better quality of experience among users.

In his thesis, Dely (2007) proposes a new hop-by-hop IP packet aggregation scheme for WMN. The concept suggests a technique of considering aggregation packets; small packets yield poor aggregation efficiency, whilst large packets are likely to be dropped when the channel quality is poor. Therefore, an optimal packet size for the channel has to be calculated; and adaptive aggregation is applied, depending on packet size. The study established that adaptive aggregation supports twice as many VoIP calls as a fixed maximum packet size aggregation. Castro *et al*. (2007) make similar recommendations, suggesting aggregating small packets for VoIP traffic. Their simulation confirmed that, while a hop-by-hop packet aggregation mechanism improves the performance of VoIP traffic in WLAN meshes, it also reduces MAC layer busy time.

Likewise, Asif *et al.* (2015) suggest a similar model known as an adaptive aggregation-based decision model (AADM), a decision-oriented dynamic solution whereby the system makes judicious aggregation decisions depending on the expected outcomes. The simulation results in the report of the study by Asif *et al.* (2015) show that AADM outperforms existing static approaches in terms of packet loss, throughput, and delay.

On a different note, preliminary investigations conducted by Zulu *et al.* (2010) reflected that header compression had the potential to increase the number of calls supported. The authors therefore, called on the need to assess how effective header compression can be in conjunction with packet aggregation on a mesh potato network.

According to Goyal *et al.* (2016), the mobile SCTP-concurrent multipath transfer (mSCTP-CMT) technique could be an alternative model. The authors discovered that mSCTP-CMT can improve the overall throughput in homogeneous networks. They therefore, propose an mSCTP-based bandwidth aggregation (mSCTP-BA) technique that enables efficient use of the resources of heterogeneous network technologies such as WLAN and UMTS. Reports from their simulation show that the mSCTP technique increases user experience and quality of service, being better by 23% for VoIP and file download applications.

Table 2.1 summarises the gaps identified in related work.

Table 2.1: Gaps identified in related work

Related Work	Gap Identified
Neves *et al.* (2018); Huang *et al.* (2019)	Field to improve the network throughput
Olariu *et al.* (2016); Asif *et al.* (2015)	Field to improve the voice call quality
Zulu et al. (2010)	Increases the number of dropped calls
Goyal et al. (2016)	mSCTP technique field to reduce the number of packet loss

2.7 Chapter Summary

This chapter provided a review of related literature. An overview and conceptual description of VOIP and VoWMN were presented. The chapter then proceeded to thematically review existing literature using the study's research objectives. Study findings from past research suggest that there is a need for the improvement of VoIP traffic in order to ameliorate communication by reducing data packet losses, jitter, and delay, among other pertinent challenges. It is upon this observation that this study is premised. In the next chapter, the methodology employed in the study is presented and justified.

CHAPTER THREE: SYSTEM DESIGN AND ARCHITECTURE

3.1 Introduction

This chapter offers the methodology used to underpin this research. In this chapter, the design of the proposed algorithm is presented, and related assumptions are provided. In Section 3.2, the proposed typical WMN system architecture is reflected. In Section 3.3, the design of the proposed algorithm and its system model is given, together with the pseudo-code. Finally, in Section 3.4, we provide a chapter summary.

This research utilised the design science research (DSR) approach. According to Carstensen *et al.* (2019), the main emphasis of DSR is on the creation and efficiency of artefacts (products), with the clear and specific goal of improving the artefacts' functionality. Figure 3.1 shows the processes that are undertaken in DSR as a research methodology.

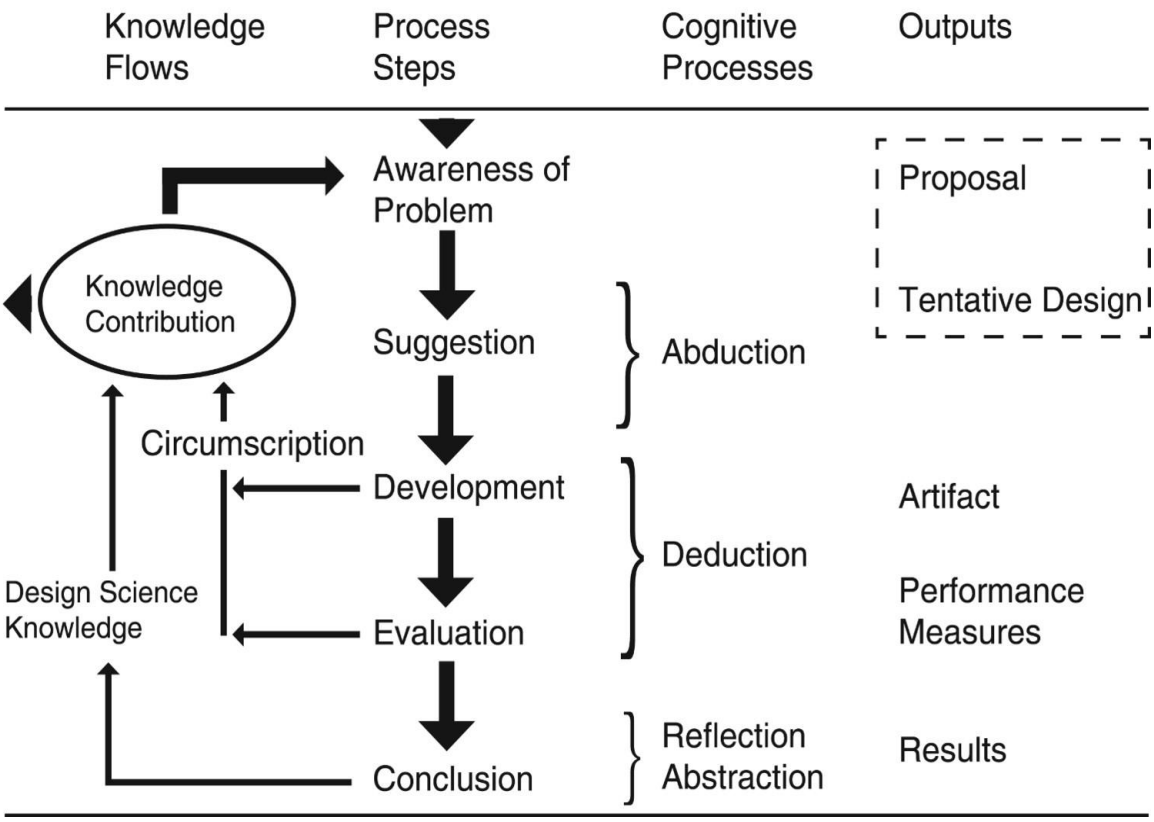

Figure 3.1 Design science processes as identified by Carstensen and Bernhard (2019)

3.2 VoWMN System Architecture

Wireless mesh networks (WMN) are communication networks made up of radio nodes arranged in a mesh topology. Mesh topology is an interconnection of all nodes in the network that are connected to one another (Parvin, 2019). Devices in the network include nodes, clients, routers, gateways, and so on.

3.2.1 Components of WMN

There are three types of nodes in WMN which are clients, routers, and gateways. The WMN clients are the end-user devices such as laptops, and smartphones that can access the network. WMN clients can then use applications such as browsing the world wide web, VoIP, playing games, location detection, and for various other tasks. WMN clients

are assumed to be mobile with limited power, having routing capability, and may or may not always be connected to the network.

The second component is WMN routers in the network to route the network traffic. These routers cannot terminate nor originate the traffic. The routers have limited mobility but reliable characteristics. Transmission power consumption in mesh routers is low for multi-hop communications strategies. Additionally, the medium access control (MAC) protocol in a mesh router supports multiple-channels and multiple interfaces to enable scalability in a multi-hop mesh environment.

WMN gateways are routers with direct access to the wired infrastructure or the internet. Because the gateways in WMNs have multiple interfaces to connect to both wired and wireless networks, they are expensive. Therefore, there are few WMN gateways in the network. Moreover, their placement has a significant impact on the performance of the network.

3.2.2 Structure of a VoIP packet

In order to offer a compression and aggregation algorithm, the researcher had to understand the structure of a VoIP packet for transmission across the network. The VoIP packet structure reflects to a great extent the hierarchical structure of the OSI. A VoIP packet has two major divisions: the header, and the payload which carries the encoded voice data. The header is made up of the internet protocol (IP) header, the user datagram protocol (UDP) header, and the real time protocol (RTP) header. These three headers constitute the header of a VoIP packet (Abualhaj *et al*., 2021). The payload or the data comes after the headers. The structure of a VoIP packet is illustrated in Figure 3.2.

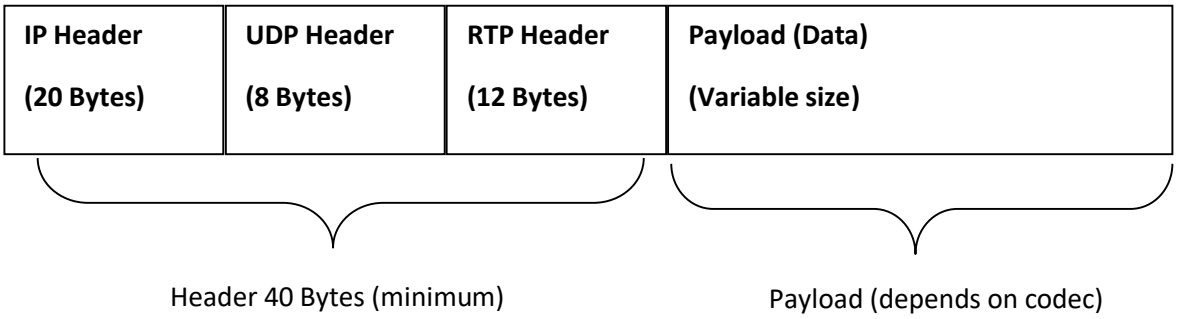

Figure 3.2 VoIP packet structure (Abualhaj *et al*., 2021)

The header is composed of three components: the IP header is 20 bytes, the UDP header 8 bytes, and the RTP header has a minimum of 12 bytes. All in all, the header is 40 bytes, minimum. The payload size depends on the codec being used.

3.3 Components of the Proposed Algorithm

The proposed algorithm will comprise two stages which will come together to create a single unit. These components are, 1) the compression of the packet, and 2) the aggregation of the packets with the determination of optimal packet size.

3.3.1 Compression of VOIP packets

There are two compression methods for VoIP packets. The first one is header compression; the other is payload compression. According to Sun *et al.* (2019), in a typical VoIP packet the payload accounts for only 33% of the total size of the packet, the rest going to the header. To save bandwidth, VoIP applications sometimes use RTP header compression. Header compression makes use of the fact that most header fields change only a little or remain static during a transmission. For example, timestamps change very little, and the source IP address does not change throughout the transmission.

Various mechanisms are available; with robust header compression (ROHC) also applicable over wireless links. ROHC, described in RFC 5225, can reduce the overhead to one byte per packet. Only the first packets contain redundant information. The following

packets contain variable information, such as identifiers or sequence numbers. These fields are also transmitted in a sufficiently compressed format to save more bits.

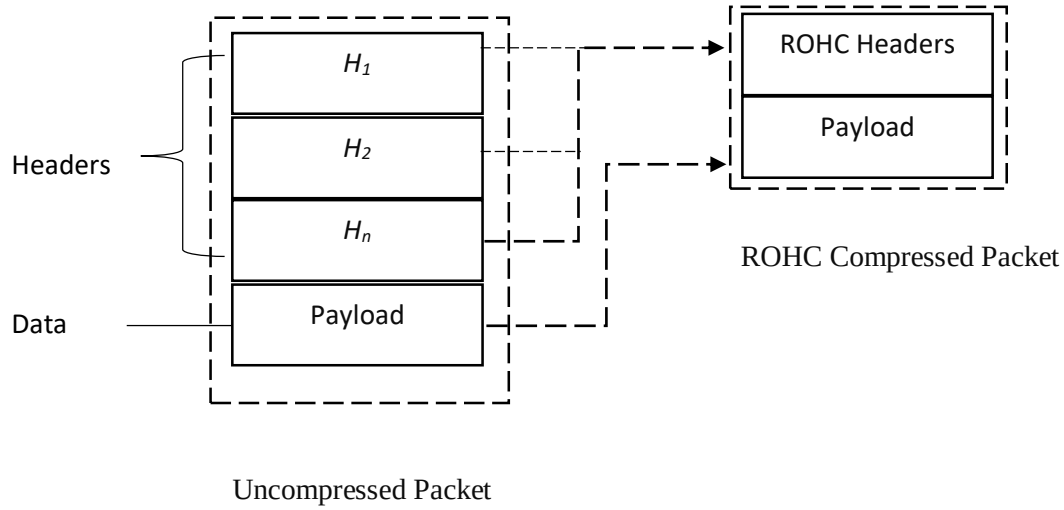

Figure 3.3: Header compression using ROHC (Sun *et al.*, 2019)

3.3.2 Aggregation of VOIP packets

Packet aggregation implies assembling one large aggregation packet from multiple small packets. The sender adds an aggregation header so that the receiver can de-aggregate the packets correctly. The packet aggregation reduces the physical and MAC layer overhead and thereby saves transmission time. Fundamentally, the packet aggregation approach works by collecting packets at a common node which is also called the aggregation target. Thereafter, the packets are merged into one large packet. This new packet is then forwarded to the destination node (de-aggregation target). At the destination node, the packets are disassembled. This is called de-aggregation. Figure 3.3 shows a high-level description of the concept.

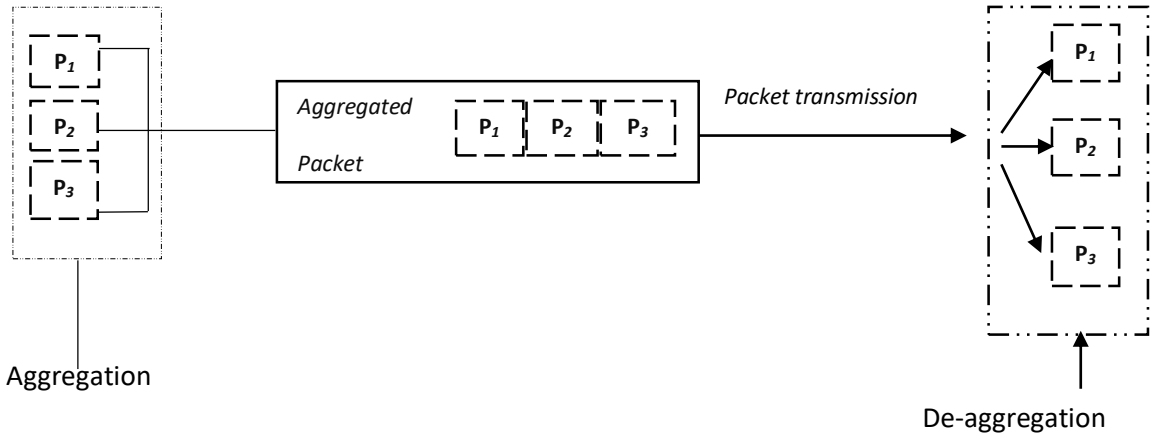

Figure 3.4 Packet aggregation concept

In Figure 3.4, three packets: P_1, P_2, P_3 are aggregated into one large packet and transmitted across a network. At the destination the packets are de-aggregated, and the receiver is given the correct packets. To aggregate packets, a node must first collect them. The outgoing stream in the MAC layer is the ideal place for achieving such. All essential data about a packet, including the IP and MAC addresses of the next node or destination node, are accessible at this point. The said queue holds packets that have been scheduled to be sent. Preferably, packets are distributed across multiple virtual queues, depending on their class of service. VoIP traffic can thus be given preference over other traffic. The natural MAC delay causes data packets to queue up if the traffic volume is high enough. However, in the event that the MAC layer is idle, some artificial delay must be introduced in low-traffic settings by not sending packets.

The next task is to aggregate the packets once they have been gathered. Outbound queue packets are aggregated based on a common aggregation destination. After aggregation, a new packet P_a is created, made up of the various packets $P_{(0\text{-}n)}$, which form the new packets' payload. The basic aggregation is illustrated in Equation (3.1) below (Abualhaj *et al*., 2021):

$$P_i + P_n \rightarrow P_{i+n} \tag{3.1}$$

As a way of identification of the various packets that make up P_a and their aggregation targets (destination node), the aggregation algorithm adds a new header H_i. When the algorithm utilizes hop-by-hop aggregation, the aggregation target for P_a is the next hop; otherwise, it remains at the end of the aggregation tunnel for end-to-end aggregation. Packet aggregation, therefore, implies that packets going to the same next hop or destination are concatenated and prepended to an extra IP header, indicating that the new packet is aggregated. On the receiving node, aggregated incoming packets are identified by the extra IP header.

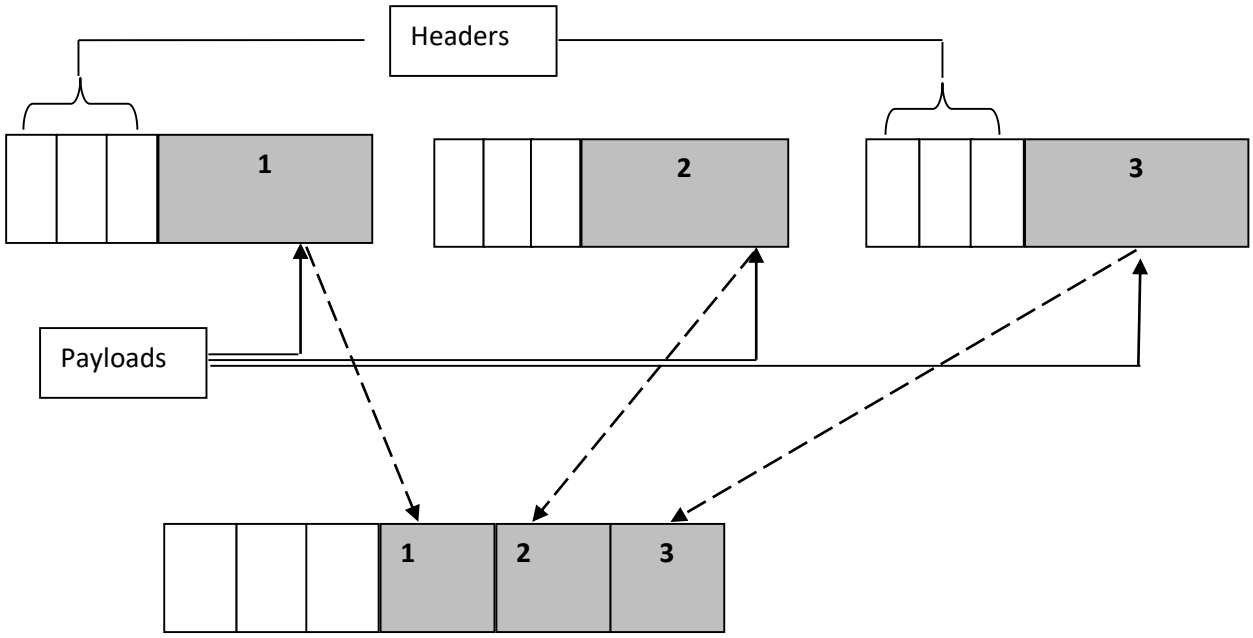

Figure 3.5 Simplified packet aggregation

Figure 3.5 shows a highly simplified packet aggregation in which the payloads of three different packets are aggregated; together with the payload of one new packet, which will be transmitted. During aggregation, new IP and MAC headers are created. The old MAC header is destroyed; however, the old IP header is retained. The old IP header cannot be discarded, because it contains the IP address of each packet, while a MAC header can

easily be replaced. The newly created IP header is responsible for holding the identification number, which is a value that each aggregated packet must have, so that the packet will be recognized in the de-aggregation module. This value is randomly selected. Figure 3.2 represents the original packet structure before aggregation, as well as the combined packets after aggregation.

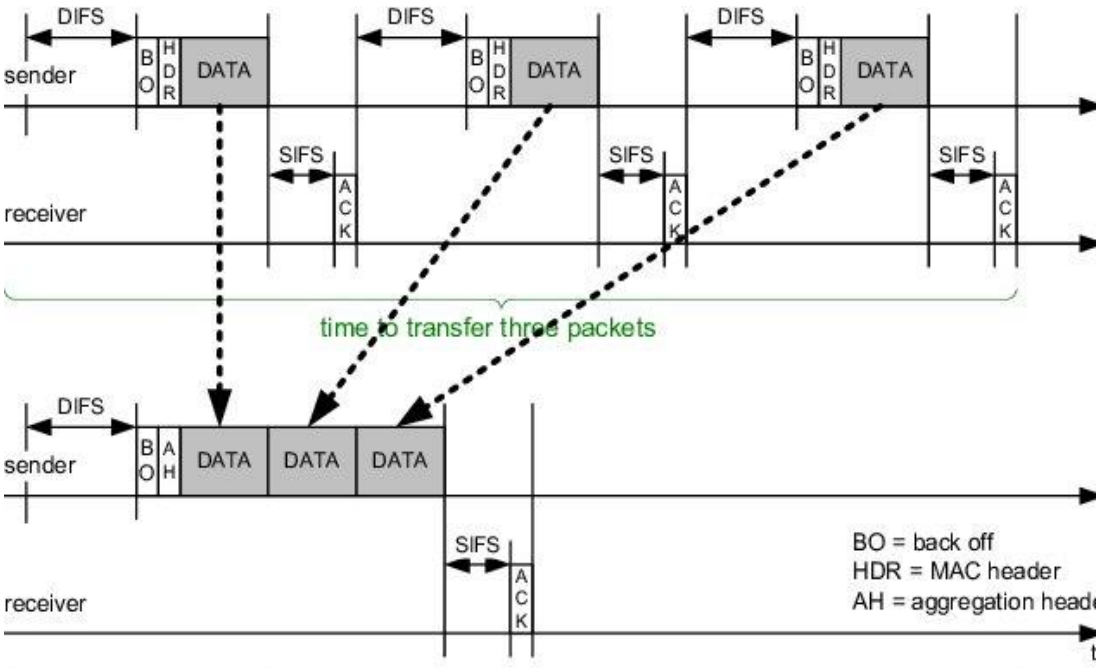

Figure 3.6 Principle of aggregation

Figure 3.6 shows a detailed illustration of packet aggregation. The diagram reflects the time saved by aggregation, and also the data savings. Initially, three different packets are shown, each with its own header DIFS, back-off algorithm (BO), and payload. On the receiver's end, each packet has its own SIFS, and gains an acknowledgment. The aggregation header redundancy is removed; only one set of BO, DIFS, SIFS, and acknowledgment were kept. Instead of the MAC header for each packet, an aggregation header is introduced. The aggregation header holds the information on the origin of each

packet. Therefore, as shown by Figure 3.6, aggregation saves time and other overheads associated with sending multiple packets across the network.

3.3.3 Determination of the optimal packet size

The concept of aggregation also includes the need for the determination of the maximum size of the packet that will be transmitted. The size of the packet to be transmitted is determined by the maximum transmission unit (MTU). The MTU is the largest packet or frame size that can be transmitted across a network without being fragmented. The maximum MTU for a VoIP packet is 1460 bytes, even though a majority of the broadband routers are set to an MTU default of 1454 bytes. Therefore, with *m* as the MTU, the limit of the packet *x* can be identified as (Sun *et al.*, 2019):

$$\lim_{x \to m} f(x) = L. \tag{3.2}$$

The easiest way of determining the optimal packet size is by pinging the destination node and gaining the response. Therefore, before packets are aggregated, there is a need to know the optimal MTU, which can be achieved by pinging and subtracting the header size from the response.

The time to wait (or aggregation delay) will be implemented using the Holding Time Packet Aggregation (Filho et al., 2012). Let A_{max} be the maximum allowed time for a packet *P* to traverse the path *P* (*s* →d), where *s* is the source node and *d* is the destination node. Node n_r is a relay node on the path *P* (*s* →d). Therefore, n_r computes the amount of time taken to reach the destination ($T_{r,d}$). As a result. n_r is able to compute the maximum holding time $H_{(p)}$, as shown in Equation (3.3):

$$H_{(p)} = \frac{A_{max} - (E_{g,r} + T_{r,d})}{|P(r,d)|} \tag{3.3}$$

where |P (*r, d*)| is the number of hops in the *P*(*r* →d) path, that is, the path from the relay node n_r to the destination node (Filho *et al*., 2012).

3.3.4 The proposed algorithm

The proposed algorithm will start by compressing packets with the same target node. At the point of origin, each packet is compressed using ROHC as indicated in Section 3.3.1. Each packet will then wait for a predetermined time to be aggregated with any other packets that might have the same destination node. If no other packet is created within the set times the packet is transmitted to the next hop on its way to the destination node. At the destination node, the packets are de-aggregated and reassembled and then decompressed. Figure 3.7 shows the principle of the algorithm. Packets are compressed and aggregated only at the source node and de-aggregated and decompressed only at the destination (target) node as in end-to-end aggregation.

Figure 3.7 provides an overview of how the proposed algorithm will work in a network. With a WMN composed of several nodes (*n0* – *nx*) and VoIP packets P_1 to P_n, with the packets originating from *n0* and *n1*, the compression will be conducted at the nodes of origin. At *n0,* there are three packets P_1, P_2, and P_3, with one packet P_n originating from node *n1*. Each packet is compressed at the source node regardless of whether or not it will be aggregated. At the source node, packets with the same destination are then aggregated, as described in Section 3.2.2. Assuming packet P_1 is created first, after compression, it will wait for the predetermined time for another packet to be created. Again, if P_2 and P_3 are created within the set time with the same destination node as P_1, the packets will be aggregated as explained in Section 3.2.2. A new aggregated packet P_a composed of P_1, P_2, and P_3 is created. Other packets will be added to *Pa*, given that P_a has not reached the maximum MTU as determined in 3.2.4; or the maximum holding time has not elapsed. That is, if no other packet arrives within the set time frame, P_a will be transmitted to the next hop in transit on the way to the destination node (*nx*). If a packet P_n is sent from a node (*n1*) and does not have either a common target node with any other packet, or no other packet is generated within the maximum holding time, P_n will be forwarded to the next hop until it reaches the destination node (*nx*). Packet P_n may go through the same node(s) as the aggregated packets, or it may traverse the network using a different path. At the destination node, all the packets are de-aggregated first to gain the original packets P_1, P_2, and P_3 from P_a. The packets are then decompressed to provide

the payloads. Because VoIP traffic is sensitive to delay, a packet that is not aggregated within the maximum delay time, such as P_n, is immediately released from the waiting queue and transmitted without being aggregated.

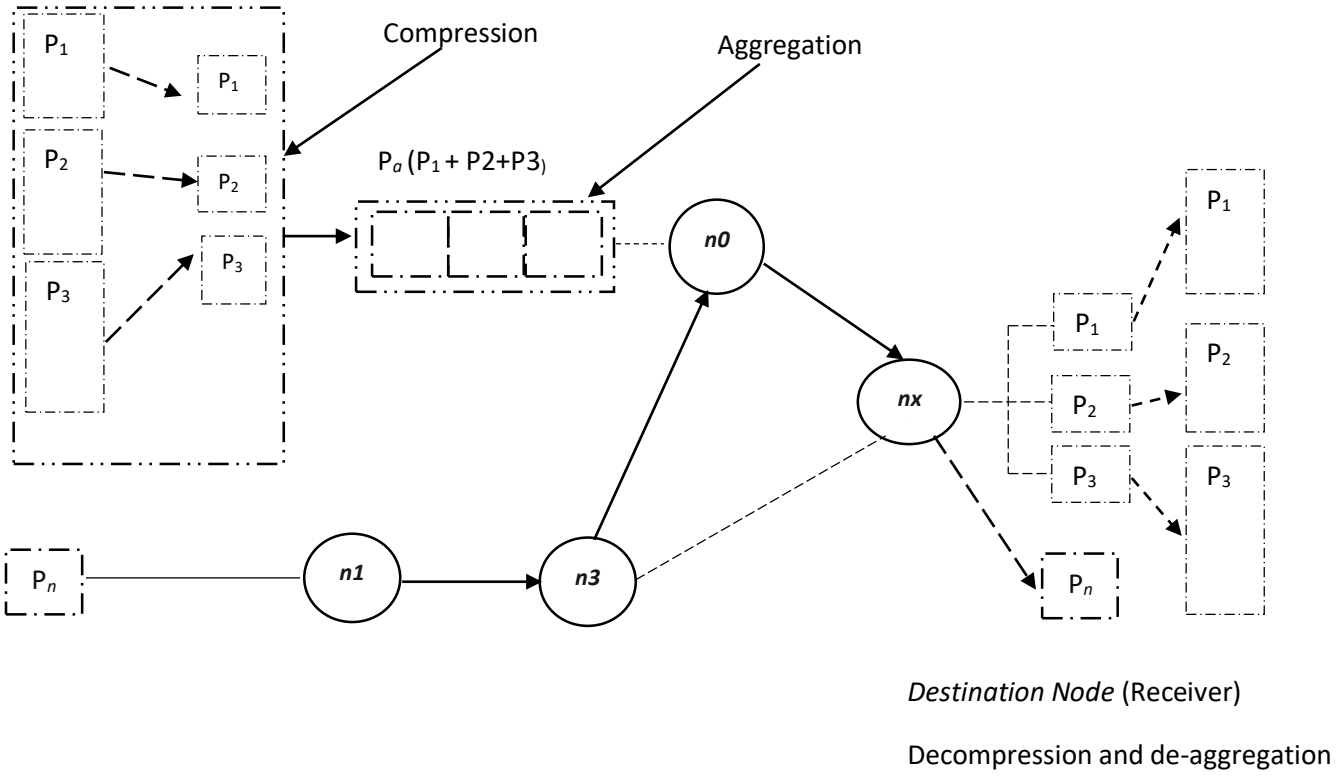

Figure 3.7 Proposed algorithm demonstration

By providing header compression, the algorithm manages not only to create smaller packets, but also maintains the VoIP quality of the original packet. If the smaller packets were then transported as they are, only half the problem would be solved. The packets would be smaller; however, the challenge of too much traffic would remain. To reduce the network traffic and increase throughput, the algorithm then aggregates the packets. This has the effect of reducing the number of packets yet delivering a larger payload. At the destination node, the packets are de-aggregated to provide the payloads.

Adaptive VOIP Optimization Algorithm

Initialise
cP //current VoIP packet
d //destination queue
Begin
cP = (*cP*+ *R*), (*cP* – (*IP, UDP*)) // Compress each current packet
if *d = ()* **then**
    d = (cP) // add current packet to the queue

or else
    while $\sum_{i=1}^{n} di \neq SIZE_{max}$ **do**
        d = d(n+cP)
        if $\sum_{i=1}^{n} di == SIZE_{max}$ then //MTU size has been reached
            drop cP
            Return
        or **else**
        if *delay(d) == MAX_{delay}* // Maximum delay time has been reached
            Return
        end if
    end while

Return P_a //one packet for transmission

Algorithm 3.1 Adaptive VOIP Optimization Algorithm

3.4 Summary

This chapter discussed the primary goal of the research, which is to implement a packet aggregation and compression algorithm. The components that make up the algorithm are discussed separately. Section 3.2.5 then provides the combined proposed algorithm. The chapter explained how each of the components of the algorithm, which are aggregation and compression will work together, and how they are integrated into one piece. The chapter also reviewed the steps that will be used in the research under the DSR approach.

CHAPTER FOUR: SYSTEM IMPLEMENTATION

4.1 Introduction

This chapter will present the implementation of the adaptive VOIP optimization algorithm described in Chapter 3. The algorithm is implemented using an IEEE 802.11s model. The model was developed per a network simulation platform NS-3 version 3.36.1. The experiment was conducted on a computer running Kali Linux version 2022.2 with kernel version 5.18. The computer had 8 GB of RAM and deployed a solid-state drive (SSD) with 100 GB of free space.

The chapter is structured as follows: In Section 4.2, an overview of the experiment set is given. The various wireless network simulation tools that were tested are also presented. In Section 4.3, the study compares various simulators used in wireless networks. The simulation setup is then provided; finally, the conclusion of the chapter is presented.

4.2 Overview of Experiment Setup

The experiments for this study were carried out using the NS-3 simulator. The NS-3 is a network simulator that models and simulates network protocols in wired, wireless, and satellite networks. The simulator employs a process known as discrete simulation. Thus, one event after another is processed, such as transmitting packets from one node to the next. The experiment does not take place in real-time. However, all occurrences are recorded on a time scale similar to actual time. The simulator includes a vast range of physical layer models, MAC protocols, routing protocols, network protocols, and application traffic generators. NS-3 is particularly beneficial for protocol comparison. Because the focus of NS-3 is on packet handling, a simulation is the exchange of packets between objects and the processing of the packets by the objects.

The simulator was run on Kali Linux (version 2022.2), an operating system (OS) that is based on the GNU/Linux kernel. Kali Linux was chosen as it is mainly used as a penetration-testing OS. Kali Linux has a plethora of tools that can be used for various

tests. Also, NS-3 is found in the Kali repositories, which makes it easier to install and maintain.

The experiments consisted of 100 VoIP packets of differing sizes (from 20 bytes to 80 bytes) injected into the network. The experiments used the iLBC VoIP codec. The packets originated from either source node, or they traversed all the possible paths. The metrics were captured for every run and the average times were computed. The experiment results indicate the average metrics for each scheme.

4.2.1 Network simulation tools

In today's fast-paced world, wireless technology is one of the areas developing at an accelerated rate. However, as a result of the uncertainty surrounding the success of new, unproven, and experimental protocols, it is not possible to implement them widely in actual computer networks. As a result, simulation tools or analytical modelling are used to test the new protocols. Should the simulation yield positive results, the protocols will be adopted and adapted. In overcoming this limitation and to calculate various network topologies without considering actual implementation, network simulation is the most popular and practical method. The scientific community makes extensive use of these to gauge novel theories and hypotheses. There are many different types of simulator; however, choosing one for a research project is crucial. Various tools such as simulators and emulators are used for this. A simulator creates an environment that mimics the behaviour and configurations of a real device. On the other hand, an emulator duplicates all the hardware and software features of a real device. Using simulation tools allows researchers to accurately represent intricacies that affect transmission rates and packet delay experienced in communication. This section provides some of the key network simulators, providing reasons for NS-3 being the chosen simulator. The simulators mentioned below are selected because of their widespread popularity.

4.2.1.1 Network Simulator-3 (NS-3)

Network Simulator-3, styled as NS-3, is an event-based open-source network simulation tool. NS-3 is a script-based simulator. According to Arvind (2016), NS-3 is primarily used

for academic and research purposes. The simulation tool is licensed under the copyleft GPLv2 License from the GNU. The simulator is used mostly in development environments for networking research. The simulator is entirely written in C++ with optional Python wrappers. What this means is that a researcher can write NS-3 simulation scripts using either C++ or Python. This makes the tool versatile and easy to use considering that one can use a high-level programming language like Python. The use of a popular programming language provides the ability for extending the simulator, particularly in terms of analysis of the metrics that will be produced by the tests Arvind (2016). NS-3 can be used for the simulation of both wired and wireless networks. NS-3 also generates PCAP packet trace files (Arvind, 2016). This enables other utilities also, such as Wireshark, to be used to analyze the packet traces.

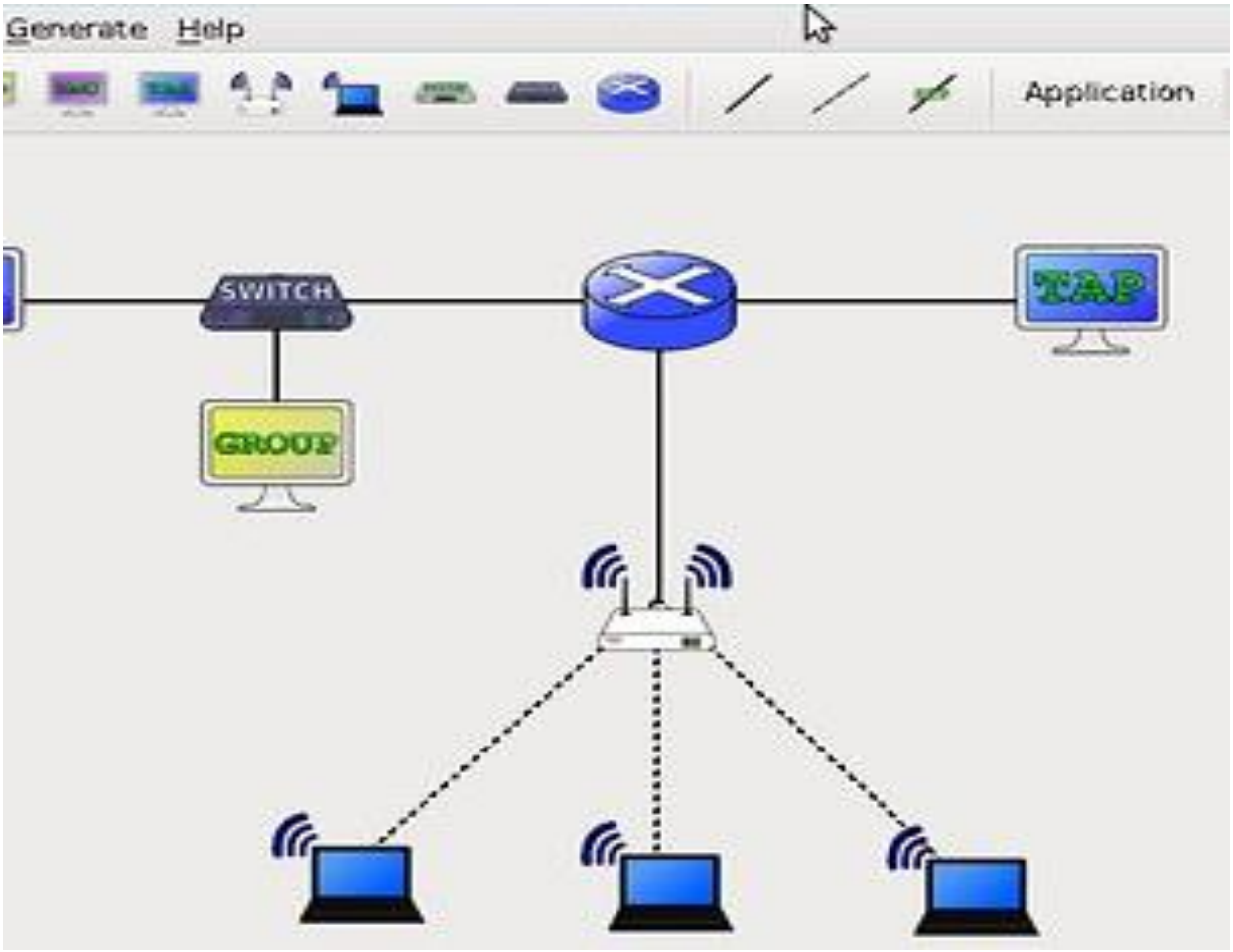

Figure 4.1: NS-3 simulator interface

NS-3 can be installed and used on a variety of operating systems such as Windows, MacOS X, Linux, and other UNIX-based operating systems, for instance FreeBSD. The simulator is easy to use, and has good documentation. However, the installation process is somewhat technical: the user downloads the simulator as source code and builds it using any C or C++ compiler. However, the documentation is highly acceptable and can assist in that regard.

4.2.1.2 OMNET++

Objective modular network tested in C++ (OMNeT++) is also a discrete event simulator similar to NS-3. The major difference is that OMNeT++ comes with its own eclipse-based integrated development environment (IDE) and a QT-based graphic user interface (GUI) used for the execution of simulations. OMNeT++ uses C++ to define the source code of protocols and applications, and has its own high-level programming language NED (network editor) to combine these implementations into modules and networks, which in turn are simulated. The simulator has very good visualization support which outranks the NS-3. It has an extensive library of predefined models, similar to the NS-3. The documentation for OMNeT++ is good, although not as good as that of the NS-3. Additionally, OMNET++ supports files from many tools such as MATLAB, Python, Perl, and R. The reason for OMNET++ not being selected is that it does not offer an undue variety of protocols; and very few protocols have been implemented.

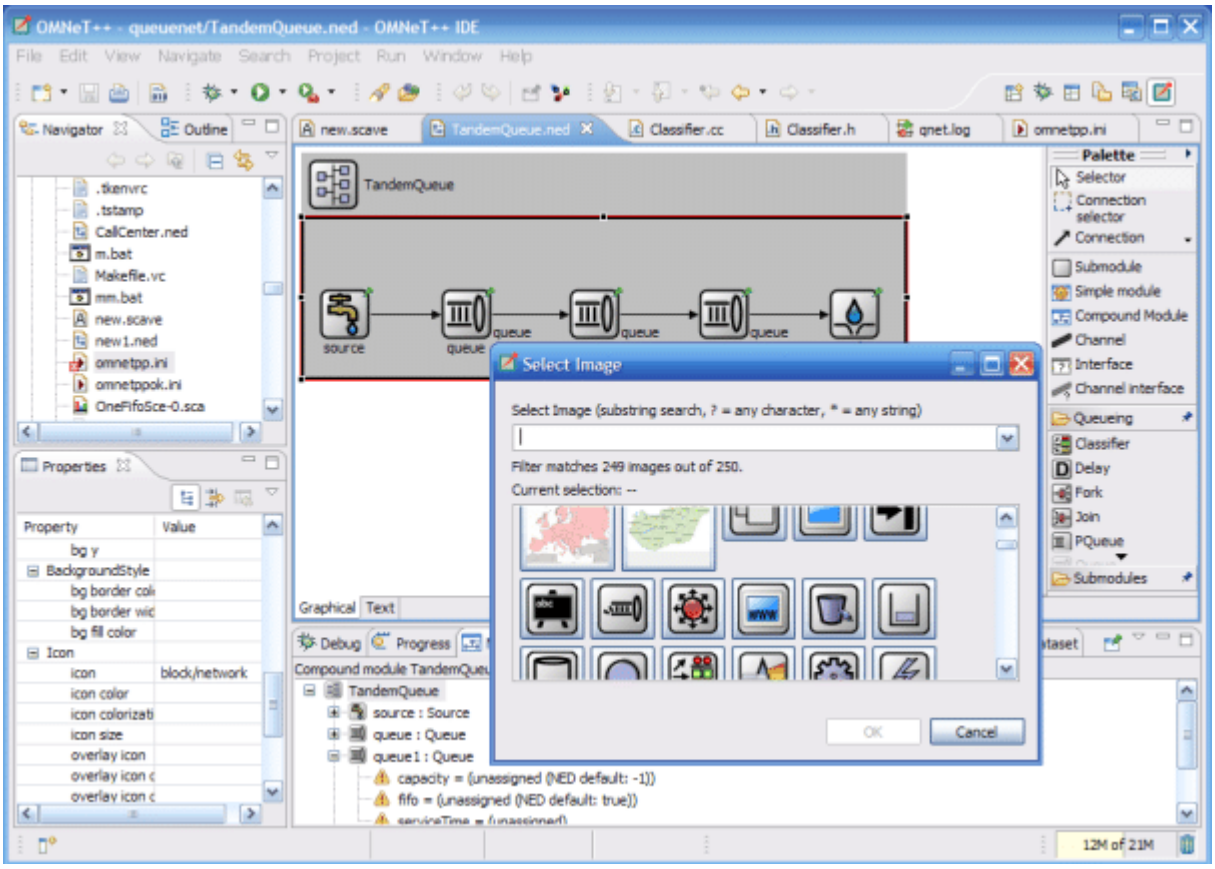

Figure 4.2: OMNeT++ simulation interface

4.2.1.3 GNS3

GNS3 or Graphical Network Simulator-3 is a one-of-a-kind network software emulator first released in the year 2008. Despite their similar names, GNS3 is not the same as NS-3. Whereas NS-3 is a script-based simulator, GNS3 emulates Cisco hardware. The key function of this emulator is to sanction a combination of real and virtual devices so that an uninterrupted functioning of complex networks can be simulated. The emulator employs emulation software known as Dynamips, a successful simulation of Cisco IOS. Unlike NS-3 or OMNeT++, GNS3 has a very high learning curve and is limited by poor implementation of wireless network simulation protocols.

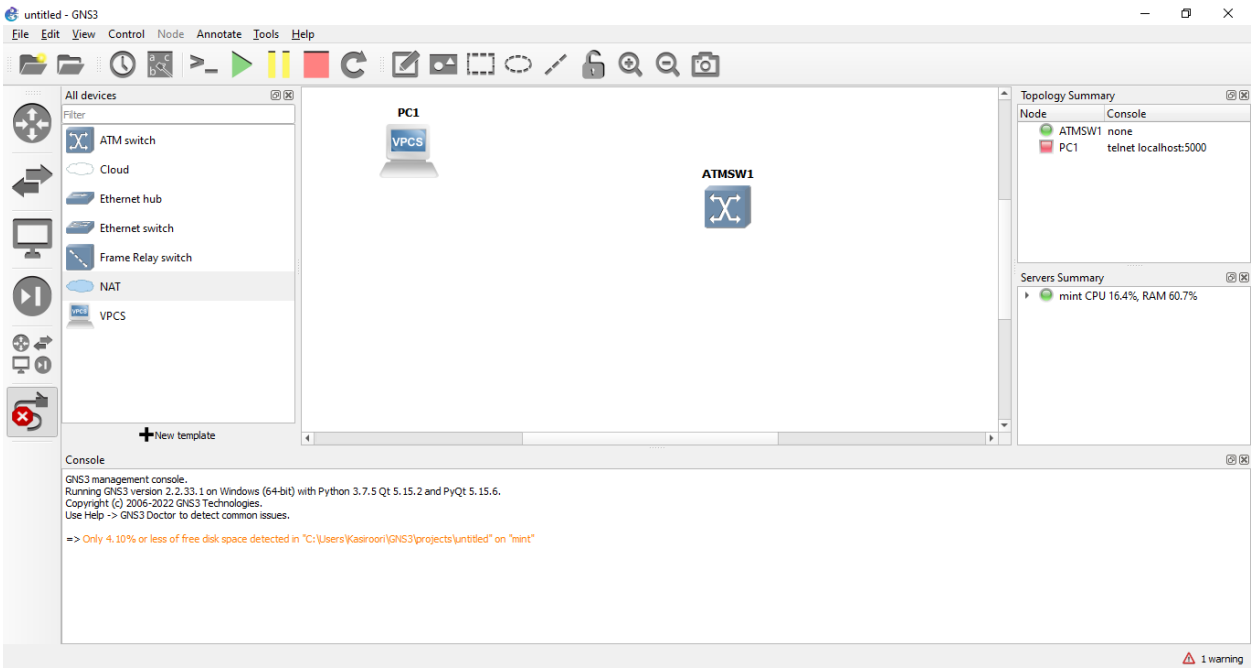

Figure 4.3: GNS3 simulation interface

The GNS3 provides a drag-and-drop interface, giving the option of running in a virtual machine (VM), on the local machine, or on a distributed cloud-based computer. It is the only tool on the list that provides an optional browser-based interface. Although GNS3 is highly intuitive and has impressive capabilities, it was not selected simply because it needed higher resource requirements than any of the simulators; and for every device added during implementation, more memory will be required.

4.2.1.4 CISCO Packet Tracer

Cisco Packet Tracer is a cross-platform visual simulation tool designed by Cisco Systems that allows users to create network topologies and to imitate modern computer networks. The software allows users to simulate the configuration of Cisco routers and switches using a simulated command line interface. Packet Tracer can be run on Linux, Microsoft Windows, and Mac OS X. There are also Android and iOS ports of packet tracer. Packet Tracer allows users to create simulated network topologies by dragging and dropping routers, switches, and various other types of network devices.

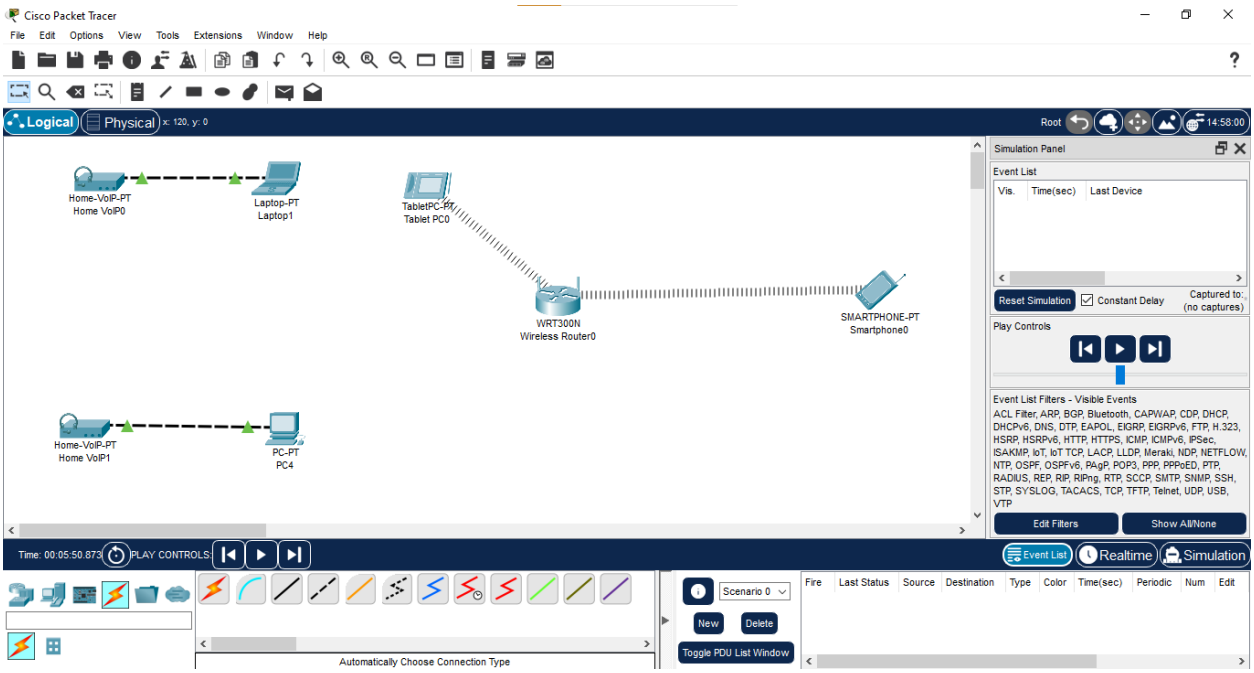

Figure 4.4: Cisco Packet Tracer interface

Packet Tracer is the easiest to use of the simulators on the list. Its drag and drop feature makes it easy for a researcher to create a simulated network. The downside of the simulator is that it can only simulate Cisco network devices. Thus, one cannot simulate a hybrid network that contains other network equipment such as Huawei devices. The simulator is also not open source, which may also limit a researcher's capability to extend or customize the software. In addition, Packet Tracer limits the number of network devices to 60 devices (Finardi, 2018). The simulator was not chosen because of the limitations.

4.2.1.5 EVE-NG

Emulated virtual environment next generation (EVE-NG) is one of the leading network simulation environments. EVE-NG is an open-source simulator. It provides two versions for download and use. The first version is known as the community version and the second one as the professional version. The community edition is completely free of charge but comes with some limitations. For example, the community version does not have the multi-user option, meaning that only one user can conduct simulations at a time. The community version also limits the number of nodes to 63 per lab, whereas the

professional version provides a limit of 1,024 nodes. The community version cannot be connected to the internet for additional simulations.

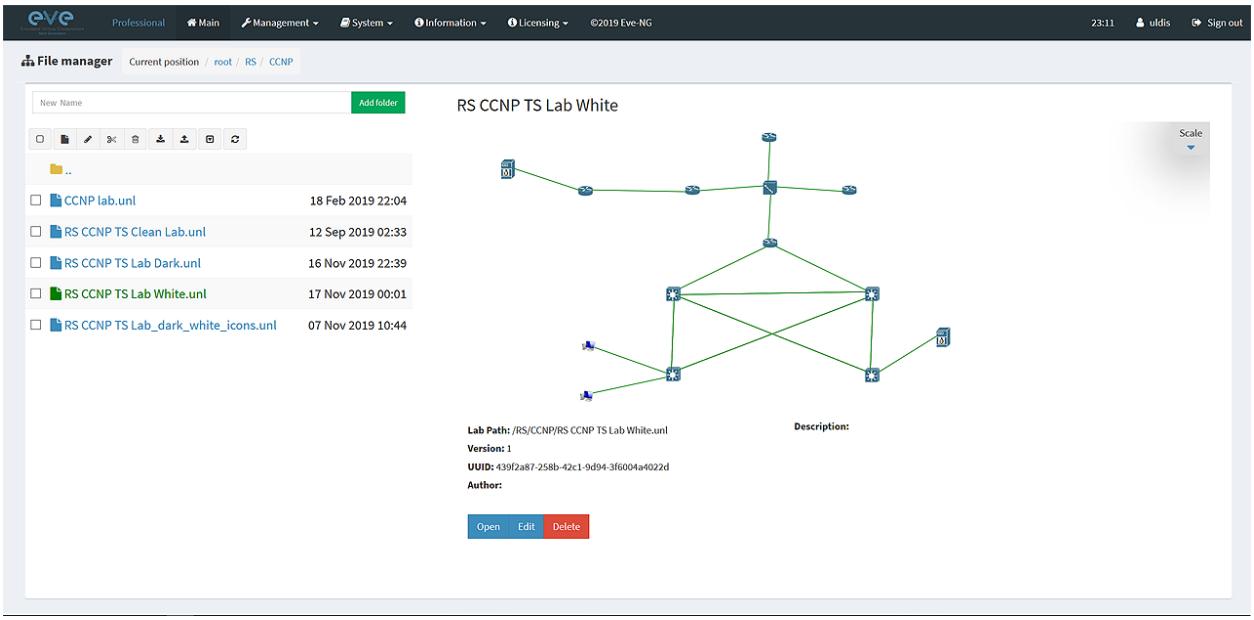

Figure 4.5 EVE-NG example simulation

The downside of EVE-NG is that it needs a VM being provided as an ISO image or an OVF (open virtualization format) file. This adds complexity to the simulator when setting it up or starting it. The download file is also the largest for the simulators with a size of 1.2 GB. In contrast, the NS-3 package has a size of 42 MB, Packet Tracer just over 200 MB, GNS3 90 MB and OMNeT++ with a size of 700 MB. Another challenge is that the use of a VM makes EVE-NG the heaviest resource user, most VMs needing around 4 GB of random-access memory (RAM). For a resource-constrained computer, this makes it slow.

4.3 Comparison of Network Simulators

This section gives a brief comparative study of the various wireless network simulators tested. The simulators are NS-3, GNS3, OMNeT++, Cisco Packet Tracer, and EVE-NG. Table 4.1 shows the comparisons. The simulations were compared on cost (license), programming language, operating systems on which they can be installed, ease of use (including learning curve), hardware requirements, and availability of documentation. All the presented network simulators are available and can be found on the internet.

Table 4.1 Comparison of Network Simulators

	NS-3	**GNS3**	**Cisco PT**	**EVE-NG**	**OMNET++**
Language	C++ and Python	Python	Java	C++	C++ (Java and C# in the IDE)
License	Open Source	Open Source	Freemium	Open Source	Commercial (Free for research)
Operating System	- UNIX-like OS + Windows	- UNIX-like OS + Windows	- UNIX-like OS + Windows	- Runs on a VM or bare metal computer	- UNIX-like OS + Windows
GUI support	Limited	Very Good	Excellent	Excellent	Good
Minimum Requirements	- 512 MB RAM - 400 MB space	- 8 GB RAM - 2 GB space	- 4 GB RAM - 2 GB space	- 8 GB RAM -20 GB space	- 1 GB RAM - 400 MB space
Documentation	Excellent	Good	Excellent. Includes video tutorials	Good	Good
Community Support	Good	Good	Good	Poor	Good

Ease of Use	Easy to use scripts Low learning curve.	Complex to use at first. High learning curve	Medium difficulty. Drag and drop GUI.	Easy to use. Drag and drop GUI. Low learning curve	Medium difficulty. High learning curve.

Legend

1. Documentation levels: Poor = Hard to understand and too technical; Good = Can be understood after going through various re-reads and using other sources; Very Good = Can be understood without need for another source; and Excellent = Written in clear non-technical language and can be used by beginners easily.
2. License: Open Source = Free to use as one pleases; Freemium = Free to use with certain conditions like subscribing to newsletter, etc.; Commercial = Needs payment to use
3. Community Support refers to non-paid support from wikis, emailing lists. etc.: Poor = Response and support rarely available on the channels; Good = Some support is available, but the community is not very active, Very Good = Support available and resolution time is fast.

4.4 Simulation Set-up of VoIP packets transfer

NS-3 was selected as the simulation tool because of the ease of use, open source, good simulation, and socket support. NS-3 also provides for PCAP output that enables analysis with other tools like Wireshark, and it has good community support. Figure 4.6 provides an overview of the NS-3 architecture. The simulation tool NS-3 was chosen mainly because of its design efficiency parameters, powerful modelling, and easy-to-implement scripts.

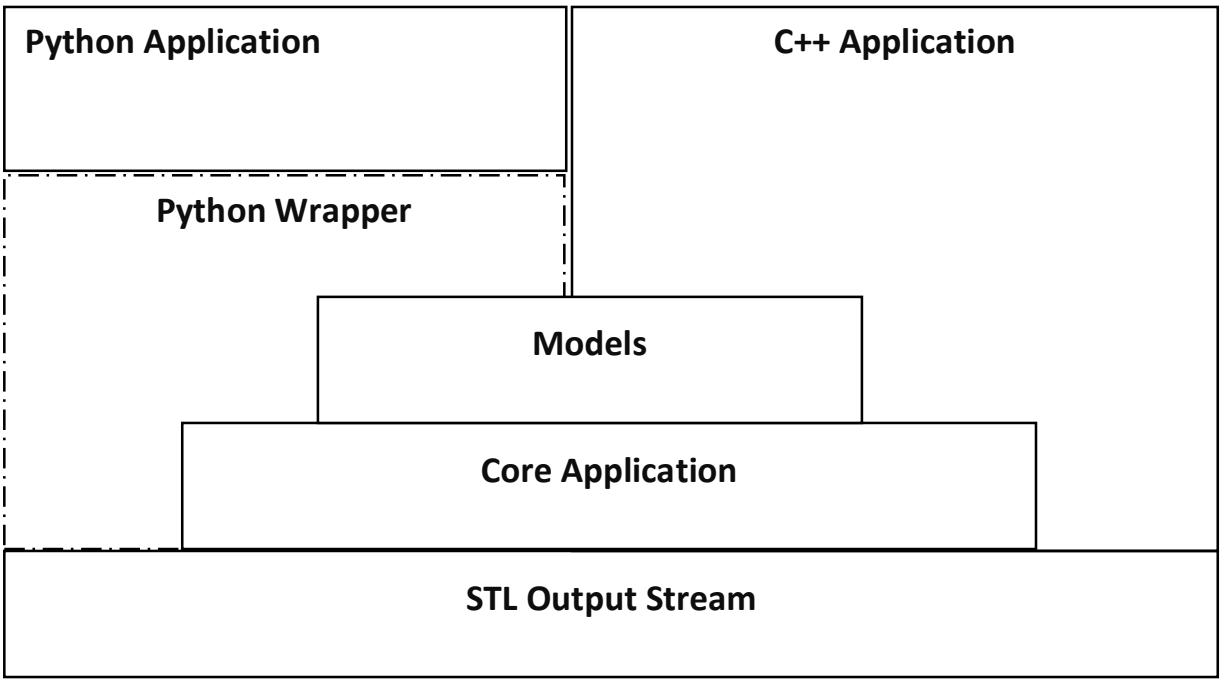

Figure 4.6: NS-3 architecture

NS-3 is written in C++. It has an output stream upon which the core simulation software is built. There are Python wrappers for the core functionality of the tool. This makes it easy for researchers to use either C++ or Python to write simulation tasks. Added to the core application are models which incorporate examples, and skeletal test scenarios which the researcher can use as templates for projects. This makes NS-3 very easy to use, even though simulation is achieved using scripts. The simulation parameters for the experiment are shown in Table 4.2, as configured in NS-3.

Table 4.2 Simulation Parameters

Simulator Tool	NS-3 version 3.36.1
Network Protocol	IEEE 802.11s
Number of Packets	100
VoIP Protocol	iLBC
Initialization	Random
Channel Type	Wireless channel
Type of Network Interface	Wireless
Number of Packets	100
Connection Radius	1 unit
Operating System	Linux (Kali Linux version 2022.2)
RAM	8 GB
Storage Type and Space	SSD with 100 GB free space

4.5 Summary

The chapter presented the simulation set-up for the adaptive VOIP optimization algorithm. The chapter provided a review of wireless network simulators, and their comparisons. It provided an explanation for NS-3 being selected as the simulator of choice, discussing the features that made it suitable for use in this research study.

CHAPTER FIVE: RESULTS AND DISCUSSIONS

5.1 Introduction

Packet aggregation and compression have emerged as critical approaches for resource conservation for WSN. For this dissertation, experiments were carried out on the proposed algorithm to evaluate its performance against aggregated, compressed, and non-aggregated packets. The outcomes of the executed schemes, as well as the numerical outcomes of each attended scenario, are presented in tables and graphs for easier study.

5.2 Test Scenarios

Table 4.1 presents an overview of the entire proposed scheme and the three existing schemes that were used in the experiment. A description of each scheme is given. The first scheme is named T1 which is the non-aggregated VOIP packet; followed by T2, in which only the packets are aggregated. The third scheme was coded as T3. This scheme used only header compression (using ROHC). The final scheme or category of packets is those named T4, which used the proposed compression and aggregation algorithm presented in Chapter 3.

Table 5.1: An Overview of the Test Scenarios (Abualhaj *et al*., 2021)

Scheme	**Test Name**	**Description of Scheme**
T1	Non-aggregated	This is a non-aggregated VOIP packet
T2	Aggregated	Packets were aggregated using end-to-end aggregation only
T3	Compressed	Only header compression was used

T4	Compressed + Aggregation	The adaptive VOIP optimization algorithm was used for these packets

5.3 Performance Parameters

This section discusses the performance parameters used to measure the effectiveness of the proposed algorithm against the existing algorithms. These are the metrics used to evaluate the efficiency of a given protocol or scheme. The performance parameters used to evaluate the design and efficiency of the schemes compared are as follows:

1. The first metric to be measured is packet loss. A packet drop occurs when packets are lost for various reasons throughout the routing process between the sender and the recipient. The packet loss for all four schemes was captured.
2. The second metric to be measured was jitter. Jitter is the variation of packet delay caused by queuing lengths, traffic, and the use of different routes throughout the network.
3. The third metric measured was latency, which is the end-to-end delay. Latency is the amount of time taken by a packet to be transmitted from a source node to the destination node. The average latency per scheme was also calculated.
4. The fourth metric measured was network throughput. This is the number of packets transmitted across the network.
5. The final metric measured was mean opinion score (MOS). The MOS measures quality of the call using packet loss rate, latency, the R-factor, and jitter. The mean opinion score listening quality (MOS-LQ) was used to measure the listening quality. The R-factor is an objective measurement based on several factors such as signal-to-noise ratio.

5.4 Performance Measurements

The various performance measurements in the experiments were captured using the *seventh.cc* data-capture module introduced in network simulator version 3 (NS-3). This module allows various data to be captured at different points of the network, with analysis

conducted by means of a familiar data-analysis package. In this study, data analysis was conducted using the Python *Pandas* package. This was made easy, NS-3 having Python wrappers for all its core functions. The metrics were the plotted using *gnuplot*.

5.5 Computer Experiments

This section describes the various experiments that were conducted and the measurements collected. The section provides a discussion of each result.

5.5.1 Results for packet loss

The packet loss was then captured for each type of scenario. The results are shown in Figure 5.1.

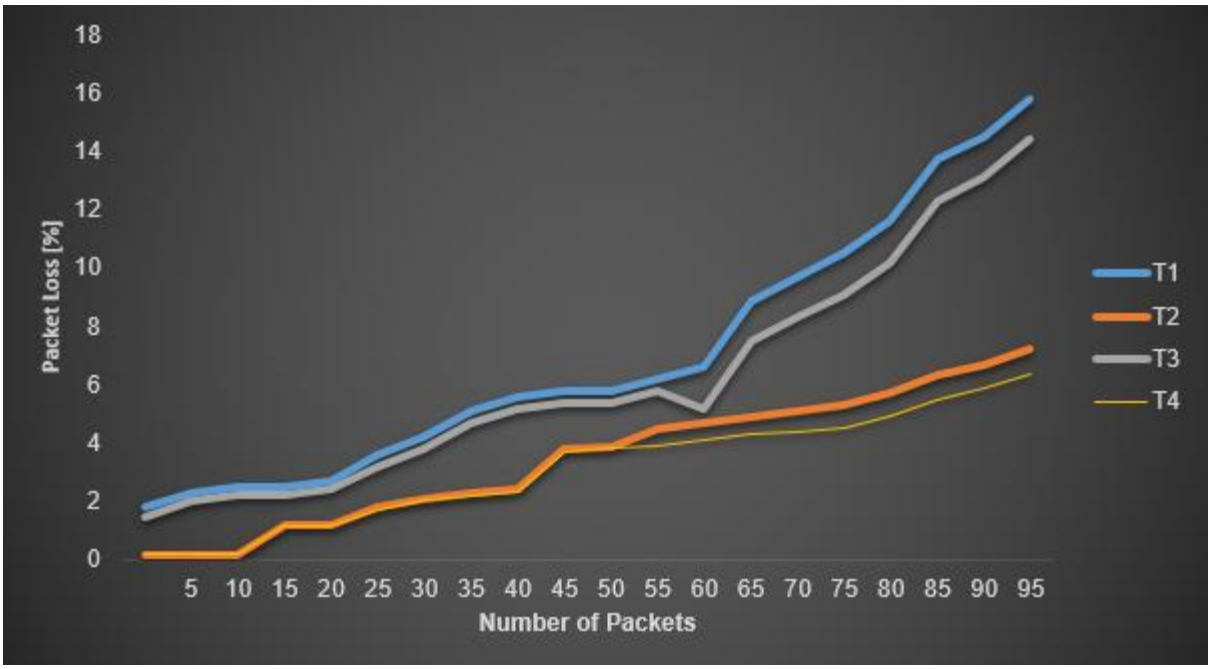

Figure 5.1: Loss of packets in non-aggregated packets

The results were captured at intervals of 5 packet injections. The experiments showed that packet losses were low up to about 20 packets, at a high of 2.7%. The graph shows an increase in the loss percentage as the number of packets increases. After about 60 packets the losses increase at a higher rate, ending at around 15.8%. Such packet losses are indicative of the network congestion due to the high traffic. When ROHC was applied

to the packets there was a noticeable difference in packet loss. Up to the 50-packet mark the difference was noticeable but not very large. However, as more packets were injected, the difference became more noticeable. Around the 60-packet mark the difference between the plain packets and those with ROHC grew from about 0.4% to more than 1.4%, an increase of over 1%. However, the greatest difference was seen when the packets were aggregated. Initially, the packet loss was near 0.2%, only sharply increasing around 45 packets to about 2.4%. The difference holds steady as the compressed packets were ultimately just over 14% or 1.4% more than the non-compressed packets. This reveals that aggregation of VoIP packets provides a substantial enhancement of the performance of VoIP packets, as shown by the T2 and T4 schemes.

5.5.2 Results for jitter

The results for the jitter tests are shown in Figure 5.2. The results of the experiment have shown that non-aggregated packets have the greatest jitter. It can be seen that, up to 15-packet injections, the jitter for the non-aggregated packets and those compressed with ROHC was almost the same, the difference being as little as 0.0076 milliseconds (ms) for most of the jitter at that time. However, from the 15-packet mark the jitter for the non-aggregated packets spiked to about 0.356ms, where additional packets only increased the jitter minimally. However, as more traffic was added, there was a corresponding rise in the jitter. The jitter growth started steadily from around 40 packets at 0.8652ms, before increasing again sharply around 70 packets, concluding at 1.2853 with 100 packets injected. By contrast, the ROHC compressed packets' jitter grew to about 0.3561ms at around 20 packets. Hence this reflects a plateau, before spiking to 0.7868ms. Thereafter, the increase grew sharply as the packets were added, ultimately arriving at 1.2664ms, almost as high as the non-aggregated packets.

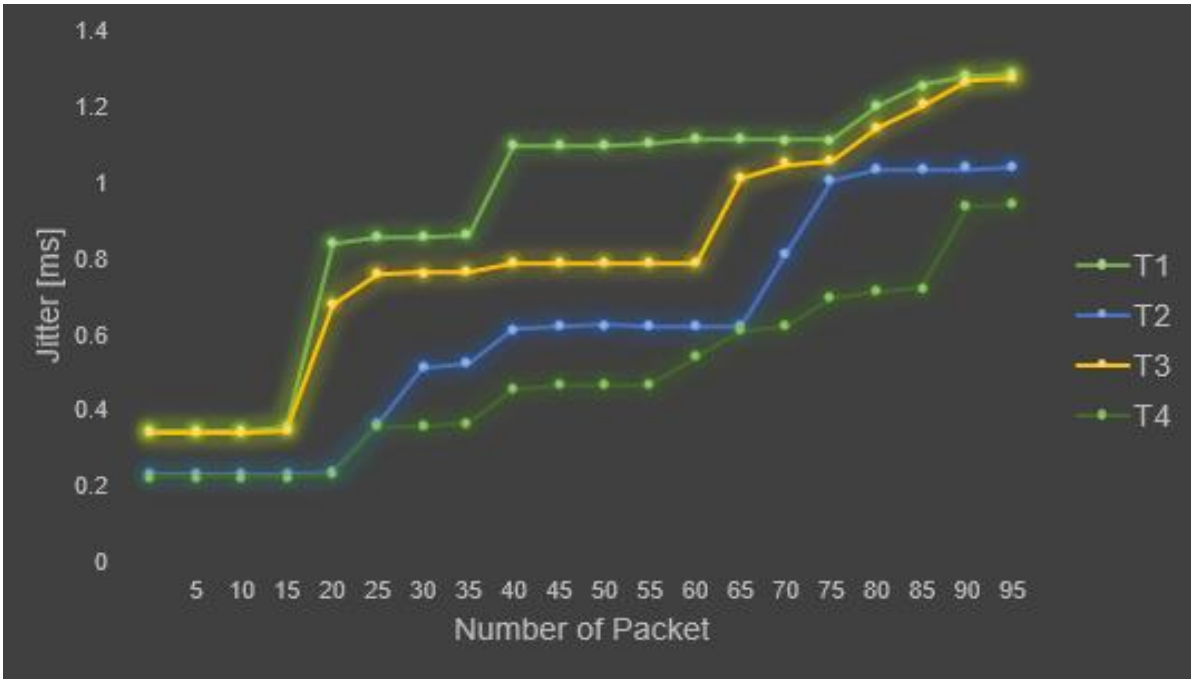

Figure 5.2: Experiment results for jitter

However, in contrast to the high jitter for the non-aggregated packets, the aggregated packets, and the compressed and aggregated packets had low jitter of around 0.3389ms to about 0.3484ms for the first 20 packets injected. Thereafter, the aggregated-only packets grew to 0.6788ms, increasing slowly to 0.7625ms before spiking sharply to 0.8125ms. Thereafter, packets increasing marginally, finally ending at 1.0378ms, or 0.2475ms lower than the non-aggregated packets. The compressed and aggregated packets (T4 scheme) always reflected the lowest jitter. However, at the 70-packet mark the jitter for the aggregated packets at 0.62435ms, and the jitter for the compressed and aggregated packets at 0.6123ms, were almost identical. The results show that there is little difference in jitter when the traffic is initially ingested. This is because the T2 and the T4 schemes which include aggregation will delay packets as the aggregation and compression takes place.

5.5.3 Results for latency

The average latency was measured and the results illustrated in Figure 5.3. End-to-end latency for each category of aggregation is shown. From the experiment, the total delay for the 100 packets utilized in the experiment may be estimated by multiplying the frame transmission delay by the quantity of frames transmitted.

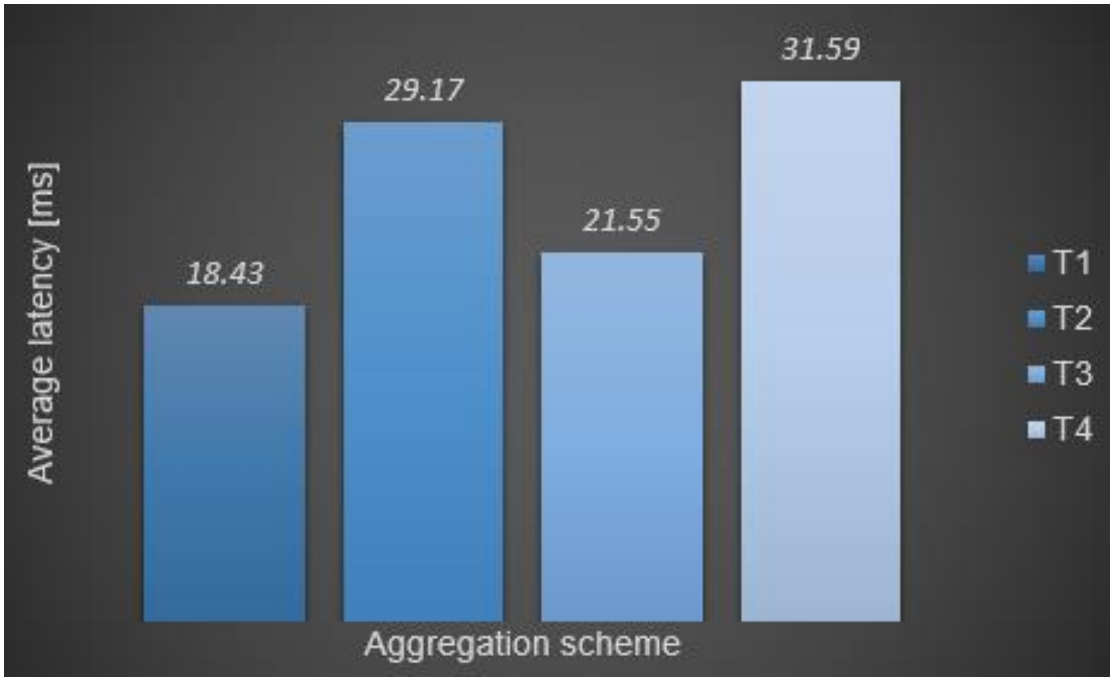

Figure 5.3: Average latency per packet scheme

The latency for the network traffic was captured. The non-aggregated packets had an average latency of 18.43ms, whereas the aggregated packets had a latency of 29.17ms. The ROHC compressed packets had average latency of 21.55ms, whilst the packets that were compressed and aggregated had the highest latency with 31.59ms. The high latency for the T4 scheme is caused by the processes that take place at the origin node in comparison with the other schemes, as shown in the algorithm. The algorithm has to identify whether there are other packets to be aggregated and/or compressed. This adds to the average latency for the aggregated packets.

5.5.4 Network throughput

The results of the throughput test are exemplified in Figure 5.4. The output was measured in megabytes per second (Mbps). The results show that much higher throughput was realized in Scheme T4 than in all the other schemes. The average throughput for this scheme was approximately 0.16Mbps. The average throughput for T1 was approximately 0.04Mbps. The T2 scheme had an average throughput of 0.12Mbps; and the last scheme, T3, had an average throughput of 0.09Mbps.

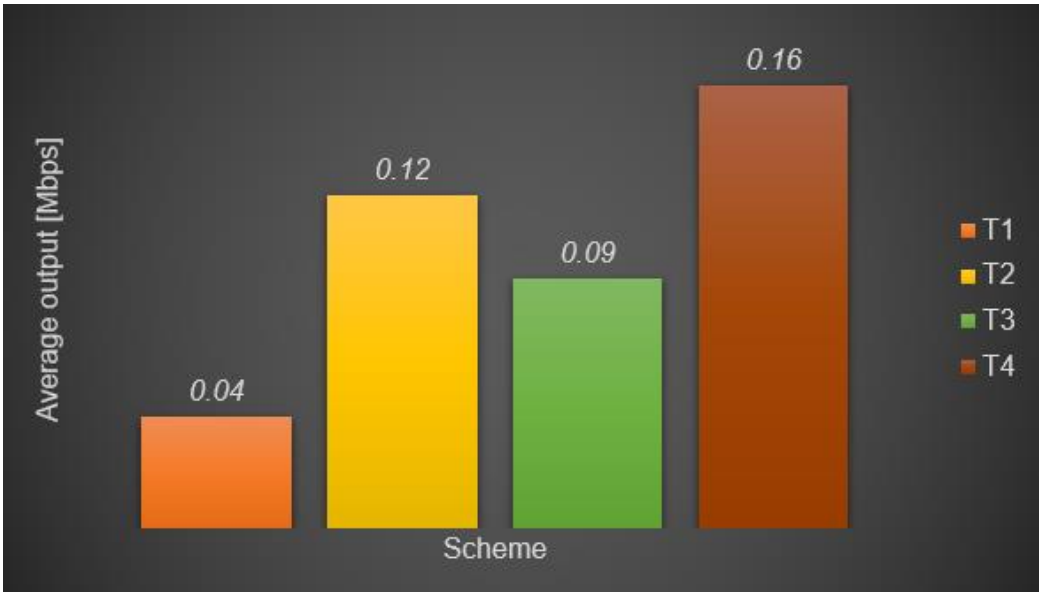

Figure 5.4: Results of network throughput

The results show that the two schemes with aggregation, T2, and T4, had a higher throughput than the other schemes. This indicates that more traffic is transmitted when the packets are compressed and aggregated.

5.5.5 Comparison of throughput against latency

A comparison of the latency and throughput averages was made to gain a better understanding of the performance of the schemes. The results are shown in Table 5.2.

Table 5.2: Comparisons of Latency vs Throughput

Scheme	T1	T2	T3	T4
Throughput (Mbps)	0.04	0.12	0.09	0.16
Latency (ms)	18.43	29.17	21.55	31.59

According to the results, the T4 scheme had the highest latency and throughput at the same time (0.16 Mbps for throughput and 31.59ms for latency). Conversely, the T1 scheme, which is non-aggregated, had the lowest throughput at 0.4Mbps. This is four times lower than the T1 scheme. The latency for the non-aggregated scheme at 18.43ms is 58.34% lower than the T4 scheme. The T3 scheme has a latency of 21.55ms which is closer to the non-aggregated scheme. However, the throughput is more than twice that of the unaggregated scheme at 0.9Mbps against 0.4 Mbps. The metrics of the T2 scheme, or the aggregated scheme, are nearer to the T4 scheme. The latency is only 2.42ms lower (at 0.12); and the throughput only 0.4Mbps less (at 0.12Mbps). The comparisons show that the T2 and T4 schemes have the best throughput, albeit with higher latency.

5.5.6 Mean opinion score results

The formula for calculating the MOS is illustrated in Figure 5.5. The calculation for MOS was conducted using the standard calculation Python programming language. The image in Figure 5.5 also shows the calculation for the T4 scheme using the formula.

```
def calculate_mos(average_latency, jitter, packet_loss):
  # Calculate Effective Latency
  # The formula takes the average latency.
  # Then jitter is added and the results are doubled
  # Then add 10 for protocol latencies

  effective_latency = (((average_latency + jitter) * 2) + 10 )
  if effective_latency < 160:
    R = 93.2 - (effective_latency/40)
  else:
    R = 93.2 - (effective_latency - 120) / 10

  # 2.5 R values per percentage of packet loss are deducted

  R = R - (packet_loss * 2.5)

  # Convert the R into an MOS value.(this is a known formula)
  mos = 1 + (0.035) * R + (.000007) * R * (R-60) * (100-R)

  return mos
```

```
print(round(calculate_mos(25.23, 0.78, 3.80), 2))
```

```
4.1
```

Figure 5.5: Calculation of MOS using Python running on Google Collab platform

The formula in Figure 5.5 is used to calculate the MOS of a given run with the average jitter, latency, and packet loss numbers. The resultant MOS is shown. The outcomes for the MOS calculation for all the schemes are shown in Table 5.3.

Table 5.3: MOS Calculation Results

Scheme	**MOS Outcome**
T1	4.534
T2	4.489
T3	4.109
T4	4.103

The findings show that the highest MOS is achieved with the T1scheme at around 4.5 followed by the T2 scheme with 4.9 and the T3 scheme with 4.109. The T4 scheme has the lowest MOS score at 4.103. According to the IUT MOS level, above 4.03 is very good.

5.6 Summary

This chapter showed the results of the experiments that were conducted using the NS-3 network simulator. The experiment parameters were also given. The results show that the T4 scheme comprising packets compressed and aggregated per the algorithm performed far better in most categories. The only test in which other schemes performed better is the latency tests.

CHAPTER SIX: CONCLUSIONS AND RECOMMENDATIONS

6.1 Introduction

In this research study, a packet aggregation and compression algorithm were designed by integrating ROHC and end-to-end aggregation, in order to improve the VoIP packet transmissions. This integration minimized the amount of packet loss, and gave a higher throughput. The algorithm was tested against some existing algorithms, performing better.

Chapter 6 is structured as follows: A discussion is provided in Section 6.2 on the way in which technical objectives were achieved. Section 6.3 provides an overall summary of the research. In Section 6.4, challenges met during this study are discussed. In Section 6.5, future work is presented; and in Section 6.6, the chapter is concluded.

6.2 Achievements of Objectives

All objectives defined in Section 1.5 of Chapter 1 have been achieved. The objectives are identified in the following passages. The major research objective shall be discussed last.

1. To review the past and current state of VoWMN

The state of VoWMN is discussed in Section 2.4. The studies reveal that, although the technology has been very useful in communication, improvements are still anticipated so that greater efficiency and effectiveness are achieved. Mesh network technology is highly dynamic and unpredictable. The review of literature pointed out that wireless mesh networks is an emerging network system suitable for various applications, including harsh environments such as military fields and emergency relief situations. However, one of the growing challenges to VoWMN is that increased volume of voice traffic in IP-based networks has presented problems of routing voice calls vis-à-vis quality, cost, and security implications. According to the literature, one of the major challenges was how to increase the network traffic without compromising on the quality of the VoIP calls. In addition, the literature showed that WMN are increasingly becoming popular for voice calls; and are replacing traditional communication systems. Research revealed that VoIP over WMN suffers from inefficient bandwidth use. Attaching a 40-byte RTP/UDP/IP header to a small

VoIP payload and 841 µs delays overhead of each packet in WMN. The authors then suggested VoIP packet multiplexing as the most appropriate solution to the problem. To address this challenge, the literature identified packet aggregation and packet compression. Considering their robustness, reliability, and speed, WMN have been adopted to enhance services in a variety of fields like broadband home network, enterprise-wide networking, metropolitan area networks, building automation, transport systems, health and medical sciences, security and surveillance systems, warehouse systems, disaster reporting, and emergency networking.

2. To design packet aggregation and compression algorithm in order to improve the performance of VoIP calls across WMN.

In Chapter 3, a discussion was provided on how the packet aggregation and compression algorithm was designed. The algorithm used a combination of ROHC and end-to-end packet compression. The algorithm included mechanisms to minimize the time a packet waits at the source node. The algorithm also has mechanisms to keep the packet size below the MTU.

The processes that take place in the algorithm are shown in Figure 3.7, and the algorithm is described in Algorithm 3.1. In Figure 3.7, the compression and aggregation, and the decompression and de-aggregation processes are identified and indicated.

3. To implement the proposed algorithm in order to increase the concurrent call across WMN.

In Chapter 4, the Adaptive VOIP Optimization Algorithm was implemented using NS-3, a network simulation software. First, five different network simulation tools were tested and compared against set criteria. The following simulators were compared in the tests: NS-3, OMNeT++, GNS3, EVE-NG, and Cisco Packet Tracer. The NS-3 simulation software was selected as it was the easiest to set up for use, had options to extend it, and used C++ and Python. The algorithm was implemented in Python.

4. To validate the proposed algorithm against existing algorithms in WMN.

The performance of the adaptive VOIP optimization algorithm was evaluated based on the following parameters, namely, packet loss, network throughput, jitter, and latency. The algorithm was compared against three existing VoIP schemes: 1) end-to-end aggregated traffic, 2) ROHC compressed-only traffic, and 3) plain traffic without aggregation or compression. The various performance measurements in the experiments were captured using the *seventh.cc* data-capture module which was introduced in NS-3. Data analysis was conducted using Python, and the metrics were plotted using *gnuplot*.

The algorithm performed better than all other algorithms in the following test categories: jitter, packet loss, and throughput. However, it had the lowest performance in latency, with the highest latency at 31.59; whilst the scheme with the lowest had 18.43, a difference of 13.16mbps. Packet loss was lower than the compared algorithms. The algorithm had only a 1.4% packet and a jitter of 0.62435ms which was also the lowest among the schemes. The adaptive VOIP optimization algorithm produced a better network throughput than did the other schemes. The MOS was calculated. The MOS measures quality of the call using packet loss rate, latency, the R-factor, and jitter. The algorithm scored 4.103 which is very good VoIP quality.

6.3 Summary of Research

The algorithm proposed in this research study minimizes the amount of packet loss and jitter, at the same time giving satisfactory throughput. The algorithm has also maintained good VoIP quality. The algorithm was designed and implemented using NS-3 simulation software, and Python as the programming language. This study developed the algorithm using compression and aggregation techniques. According to the presented results, the proposed algorithm increases the number of packets (traffic) that can be transmitted at a given time by as much as one-and-a-half times against other algorithms. The study showed that the algorithm maintained good VoIP quality compared with the other algorithms.

6.4 Encountered Challenges

The greatest challenge was testing the various network simulators and creating the test scenarios. The learning curves for the simulators added to the complexity of the research. For simple test cases it is recommended that researchers target simulators that are GUI based, as they have a lower learning curve. However, for complex text scenarios, researchers can use simulators that allow for extended modules, using a language they are familiar with. This eases the adoption challenges.

6.5 Future Work

This study sought to establish the feasibility of using both compression and aggregation to increase the number of calls across WMN when transmitting VoIP. The study did not review any potential security issues that may have been introduced by combining the two techniques. This can be an area for further research.

6.6 Summary

This chapter concluded that the adaptive VOIP optimization algorithm attained its objective of improving the VoIP transmission in WMNS. The algorithm was able to achieve this through minimizing packet loss, increasing throughput, minimizing jitter, and maintaining good VoIP quality. The algorithm performed better than the non-aggregated and non-compressed traffic. Since the algorithm managed to achieve all the defined goals, it is considered that the ultimate purpose of this research study has been successfully achieved.

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