I. Performance Optimization of concurrent VOIP calls across Wireless Mesh Networks

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***Abstract*—Over the past years we have seen an increasing number of mobile users with the need and desire for optimized concurrent VOIP calls across Wireless Mesh Networks (WMN). Thus, this study sought to develop a packet aggregation and compression algorithm in order to improve the performance of VoIP calls across WMN. The algorithm was designed by integrating ROHC and end-to-end aggregation. This integration minimised the amount of packet loss and gave a higher throughput. The algorithm was tested against some existing algorithms and performed better.**

Keywords—VoIP, VoWMN, packet aggregation, packet compression

II. Introduction

Humanity has always found the use of voice as an essential way of communicating. Thus, there mankind has always looked for ways to enhance the use of voice in everyday communication. The creation of the public service telephone network (PSTN) was one such endeavour which provided real-time communication with other people located far. As technology improved, voice over IP (VoIP), a technology that enables one to make voice calls utilising a broadband Internet connection instead of a traditional (or analog) phone line was developed. VoIP is frequently selected over other options like PSTNs [1][2]. IEE 802.11 identify Wireless Mesh Networks (WMN) as one of the emerging types of networks technology that are highly available and compatible with VoIP at the same making it easy to deploy VoIP where delivery of voice communication is difficult and reducing costs at the same time [3].

The implementation of VoIP necessitates a more robust network infrastructure, thus, WMN, a network infrastructure technology, was created to enhance VoIP's coverage and performance [4]. The technology still has challenges and voice codecs, jitter, delay, and packet loss as having an impact on the quantity or quality of VoIP calls across WMN [5]. VoIP over WMN studies, have mainly focused on packet aggregation as a way to mitigate the challenges whilst a small number combined some aggregation and compression algorithms, but none of them used the WMN with a high performance standard like 802.11ac. IEE 802.11 includes a number of standards, including 802.11b, g, n, and s, that can be used as WMN standards. Due to its high bandwidth capacity, 5GHz frequency, and up to 7GB throughput, this study focuses primarily on the utilisation of 802.11ac [6]. Thus, the goal of this research was to create simulations that would enhance the performance of VoIP over WMN in an 802.11ac scenario by finding a technique to increase the number of concurrent calls while reducing the impact of jitter, delay, and packet drop.

III. Related Work

This section will provide a synthesis of related work as known as a literature review. A literature review helps the researcher to establish how the current study is related to those carried out in the past studies. The literature review will cover the following themes: 1) Overview of VoWMN and VOIP, 2) WMN and VoIP routing protocols, 3) Current state of VoWMN.

A. Overview of VoWMN and VOIP

VOIP is one technological advancement that man created and is constantly perfecting to ensure that voice communication stays the cornerstone of human communication. In contrast to PSTN, which offers dedicated end-to-end circuit connections for the duration of each conversation, VoIP systems enable real-time transmission of voice signals in the form of data packets across Internet Protocol (IP) networks. Due to its utility in communication, VoIP has become a very popular technology [7][8][9] spanning a host of popular VoIP technologies or applications like Skype, Viber, WhatsApp and a variety of other technologies with VoIP capacity. According to [2] VoIP technology has become more widely used since it provides the highest degree of service quality, is reasonably priced, and is more reliable.

Despite the fact that VoIP is beneficial it is vital to note that there are a few issues with it that need to be resolved if further advantages of using VOIP are to be realised. The technology faces some difficulties, such as network or bandwidth capacity, network architecture, system design, performance, reliability, availability, scalability, security, regulatory constraints, and issues with quality of service [8][10][11][12].

VoWMN are a very attractive way to extend the network coverage into the dead zone in cases where the wired network is not easy to install and for enterprise infrastructures [13][14][15]. WMN is a multi-hop wireless network made up of

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communication nodes following a mesh topology [3] Additionally, [16][17] point out that WMN help to enable individuals to be continuously online thereby solving problems associated with wired networks. Mesh networks possess a comparative advantage over wired LANs because of ease of deployment ease of expansion, better coverage, robust to node failure and reduced cost of maintenance [15][18]. 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 others for backhaul connection [15] and a typical VoWMN network architecture is shown in Fig 1.

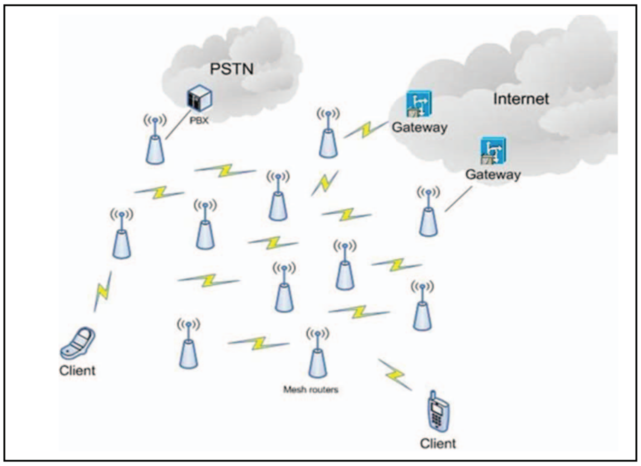


Fig. 1. Typical VoWMN network architecture

B. WMN and VoIP routing protocols

In VoWMN, node mobility which is a key trait, can increase packet loss, latency, and jitter. To improve VoWMN efficiency, [19] suggests that VOIP solutions in WMNs should feature mobile nodes that enable background traffic, raising quality of service standards and guaranteeing packet data delivery. According to [19] by determining the integration options and including supportive mesh nodes, VoIP QOS in WMN can be improved. Using the suggested methods by [19] results in an improvement of VoIP quality on a 5-point MOS rating-scale of 0.2 in scenarios with no mobility, 2.2 in partial mobility, and 0.9 in full mobility. Researcher [21] suggests using cloud-based systems with mechanisms to deal with various objective preferences, workloads, and cloud features as an alternate approach to allowing cost-quality optimization of VOIP systems.

Additionally, [22] suggested an adaptive informant factor (AIF) model based on optimal channel allocation (OCA) that reacts anytime interference is detected. Due to constant monitoring and maintenance, responding to interferences that are discovered guarantees that packet transmission maintains its optimal level.

Simulations involving both aggregation and compression algorithms with advanced routing protocol are discussed in eight of the sources that were reviewed. A data aggregation back pressure routing (DABPR) scheme is proposed by [23]. In the scheme overlapping routes are instantaneously aggregated for effective data transmission and to prolong the network's life. The routing algorithm includes cluster-head selection, maximization of event detection reliability, data aggregation, scheduling, and route selection. Reference [24] 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 which enables any cluster to operate a batch without necessarily setting up overhead. In contrast [25] preferred a compressed fuzzy logic based multi-criteria Ad hoc on demand distance vector (AODV) whereby routing decisions are dependent on number of relays, distance factor, direction angle, and vehicles speed variance. An earlier study by [26] presented nearly similar recommendations. A QoS-aware routing protocol with adaptive feedback scheme for video streaming for mobile networks is advanced by [27]. The protocol has mechanisms for detecting link failures in a route in order to re-establish connections so that quality of service is maintained. Reference [28] is of the opinion that a load balancing ad hoc on-demand multipath distance vector (LBAOMDV) routing protocol is more appropriate as it regulates the fair usage of both node energy and available bandwidth by exploiting the availability of multiple paths for data transfer. Finally, [29] recommends an energy-efficient routing algorithm that strives to strike a balance between data traffic among the nodes and network lifetime by the use of Shortest Path Tree (SPT) and Minimum Spanning Tree (MST) under the auspices of Distributed Energy Balanced Routing (DEBR) and Shortest Path Aggregation Tree Based Routing Protocol.

C. State of VoWMN

Researchers note that WMNs are a new type of network system that can be used in a variety of difficult contexts, such as military settings and disaster relief efforts [11][19] and using VoIP on WMNs still has challenges like inefficient bandwidth utilisation, for example attaching a 40-byte header to a short payload and an 841µs delay overhead per packet according to [36] and [37].

Various ways to improve the capacity of VoWMN have been proposed. Researcher [38] suggest a classification method which categorises VoIP packets as either high priority or low priority, with high priority packets receiving preference over those in the low priority category using a dynamic optimisation algorithm that identifies the ideal subset of high priority speech segments [35]. A Delay-aware Packet Prioritization Mechanism (DPPM) is suggested by [5] with the intention of evenly distributing the Quality of Service (QoS) levels across all VoIP calls in WMNs. The method prioritises VoIP packets that have been in the queue for a longer period of time across many hops within the WMN improving voice call quality and capacity [5].

Another technique that has been identified as a way to increase WMN's efficiency is known as packet aggregation [33]. Holding Time Aggregation (HTA) by [39] uses an adaptable packet retention time to let relay nodes explore aggregation opportunities on a multi-hop path, keeping jitter and latency within set application limitations.

A study by [40] reduces transmission volume and communication overhead without adding additional latency by separating each packet into forwarding packets and aggregating packets. According to [40] the algorithm reduced latency by 31.33% to 51.41%. The adaptive aggregation-based decision model (AADM) proposed by [32] is a dynamic decision-oriented solution in which the system makes aggregation decisions based on the anticipated outcomes and simulation results show that in terms of packet loss, throughput, and delay, AADM performs better than current static techniques [32], allowing twice as many VoIP calls as a fixed maximum packet size aggregation [30][31].

Another technique used to improve VoWMN efficiency is header compression. It has the ability to enhance the number of calls supported, as [33] demonstrated. Reference [34] suggested the use of a mobile SCTP-Concurrent Multipath Transfer (mSCTP-CMT) based Bandwidth Aggregation (msctp-BA) technique to boost throughput over different network technologies including WLAN and UMTS. The results of their simulation demonstrate that the mSCTP technology VoIP quality of service since by over 23%.

IV. Methodology

The study employed the Design Science Research methodology which is defined by [41] as a problem-solving paradigm that seeks to enhance human knowledge via the creation of innovative artifacts. Using this methodology, an innovative, purposeful artifact for a VoIP traffic transmission was created in order to address the problem identified. In addition, the designed artifact was evaluated in order to ensure its utility for the VoIP transmission problem. This was necessary because the researcher wanted to ensure that the purpose of the study had been met. In order to ascertain novelty and contribution to research, the researcher tested the artifact's ability to solve the identified VoIP problem by providing a more effective solution when compared to earlier attempts to solve the same problem.

A. *System Design And Architecture*

The design of the proposed algorithm is presented and related assumptions are provided. The proposed typical WMN system architecture is also presented.

B. *Structure of a VoIP Packet*

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 which has 20 bytes, then the User Datagram Protocol (UDP) header with 8 bytes and then the Real Time Protocol (RTP) header with a minimum of 12 bytes. Those three headers constitute the VoIP packet header which is 40 bytes minimum. The payload or the data which comes after the header depends on the codec being used.

C. *Packet Aggregation*

Aggregation refers to the process of creating a single, substantial packet from several smaller ones. To ensure that the recipient can correctly de-aggregate the packets, the sender provides an aggregation header. Fundamentally, the packet aggregation strategy collects packets at a single node, also known as the aggregation target, merges the packets into a single large packet, and then sends this new packet to the destination node, also known as the de-aggregation target, where the packets are dismantled in a procedure known as de-aggregation. This basic idea is depicted in Fig 2.

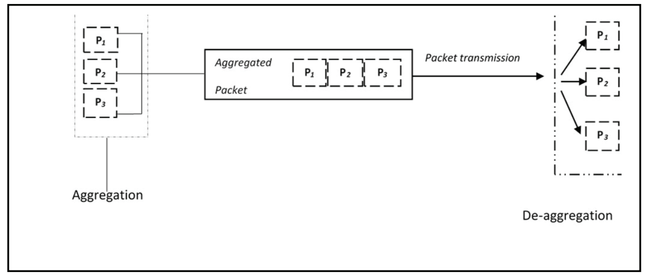


Fig. 2. Packet Aggregation concept

One large packet P_a made up of all the packets $P_{(0-n)}$ is sent over the network. Packet collection and the creation of new IP and MAC headers is done at the MAC layer since it provides access to all pertinent information about a packet, including the IP and MAC addresses of the next node or destination node. While the old IP header is maintained, the old MAC header is destroyed. Unlike a MAC header, which can be easily changed, the old IP header cannot be removed because it carries the IP address of every packet. Each aggregated packet needs to have an identification number, randomly selected, which the newly generated IP header is in charge of storing, in order for the packet to be recognized by the disaggregation module. The basic aggregation is described in (1).

$$P_i + P_n \to P_{i+n} \tag{1}$$

Aggregation's cost savings are made since distinct packets each with a distinct header, back-off method (BO), and payload, SIFS and receives an acknowledgement at the receiver's end. Aggregation eliminates redundant headers, as just one set of BO, DIFS, SIFS, and acknowledgement is retained. An aggregation header is used in place of the individual packet's MAC header therefore sending fewer packets over the network.

The necessity for determining the maximum size of the packet that will be transmitted is part of the concept of aggregation. The Maximum Transmission Unit (MTU) determines the maximum size of the packet that will be transmitted across a network without being fragmented. Most broadband routers have an MTU default setting of 1454 bytes for *VoIP* transmission. Equation (2) can be used to determine the limit of the packet *x* where *m* is the MTU.

$$\lim_{x \to m} f(x) = L. \tag{2}$$

The time to wait (or aggregation delay) will be implemented using the Holding Time Packet Aggregation [42] as described in (3). Let A_{max} be the maximum allowed time for a packet P to traverse the path $P(s \to d)$, where s is the source node and d is the destination node. Node n_r is a relay node on the path $P(s \to d)$. Therefore, n_r computes the amount of time 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 (3).

$$H_{(p)} = \frac{A_{max} - (E_{g,r} + T_{r,d})}{|P(r,d)|} \quad (3)$$

Where $|P(r, d)|$ is the number of hops in the $P(r \rightarrow d)$ path, that is, the path from the relay node n_r to the destination node [42].

D. Packet Compression

There are two compression methods for VoIP packets. The first one is header compression and the other one is payload compression. According to [43] in a typical VoIP packet the pay load accounts for only 33% of the total size of the packet with 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 only change little or stay static during a transmission. For example, timestamps change very little, and the source IP address does not change throughout the transmission.

Different mechanisms are available, whereas Robust Header Compression (ROHC) is 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 [44]. The following packets only contain variable information, such as identifiers or sequence numbers. These fields are also transmitted in a sufficiently compressed format to save some more bits.

E. The Adaptive VOIP Optimization Algorithm

The proposed algorithm will be composed of two processes which will come together to create a single output. These components are 1) the compression of the packet and 2) the aggregation of the compressed packets.

Fig 3 illustrates how the algorithm will work in a network and Algorithm (1) describes the proposed algorithm including the management of delays and maximum size of packet transmissions. The compression will be done at the nodes of origin and each packet is compressed at the source node regardless of whether it will be aggregated or not. Packets with the same destination are then aggregated. A new aggregated packet Pa composed of the original packets is created. Each packet will then wait for a predetermined time calculated using HAT, to be aggregated with any other packets that might have the same destination node. If the packet Pa has not reached the MTU or the maximum holding time has not elapsed, other packets will added otherwise Pa will be transmitted onwards to the destination node.

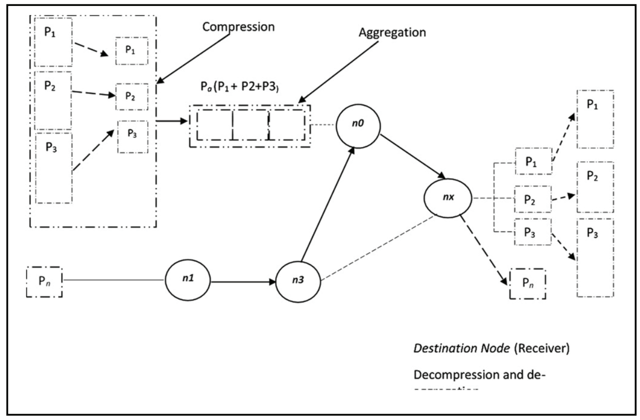


Fig. 3. Proposed algorithmn in a network

At the destination node all the packets are de-aggregated first to get the original packets from Pa. The packets are then decompressed to provide the payloads. Since VoIP traffic is sensitive to delay, a packet which does not get aggregated within the maximum delay time is immediately released from the waiting queue and transmitted without being aggregated and is transmitted across a network without being fragmented. The algorithm uses en-to-end aggregation and packets are only aggregated at source node.

Adaptive VOIP Optimization Algorithm

Initialise
cP //current VoIP packet
d //destination queue
Begin
$cP = (cP + R), (cP - (IP, UDP))$//Compress each packet
if $d = ()$ **then**
$d = (cP)$ // add current packet to the queue
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 reached
drop cP
Return
else
if $delay(d) == MAX_{delay}$ // Max delay time reached
Return
end if
end while
Return P_a //one packet for transmission

Algorithm 1. Adaptive VOIP Optimization Algorithm

The algorithm is able to both build smaller packets and preserve the VoIP quality of the original packet by applying header compression and aggregation simultaneously This also reduces network traffic and increases throughput resulting in fewer packets being sent, but a greater payload is delivered. Only half of the issue would be resolved if the smaller packets were delivered in their current state. Although the packets would be smaller, the problem of excessive traffic would still exist.

V. FINDINGS AND DISCUSSIONS

Experiments were carried out on the proposed algorithm to evaluate its performance against aggregated, compressed, and non-aggregated packets.

A. Test Parameters

Table I presents an overview of all the proposed scheme and the three existing schemes that were used in the experiment. The schemes were named TP1, TP2, TP3 and TP4. The description (explanations) for all the schemes along with the test names are provided in Table I.

TABLE I. OVERVIEW OF THE TEST SCENARIOS

Test Scenarios		
Test Name	***Scheme***	***Description of scheme***
TP1	Non-aggregated	Non-aggregated VOIP packets
TP2	Aggregated	End-to-end aggregated packets
TP3	Compressed	ROHC Compresses Packets
TP4	Compressed & Aggregated	The Adaptive VOIP Optimization Algorithm was used for these packets

The performance parameters used to evaluate the design and efficiency of the schemes we compared are as follows. Packet loss, jitter, latency, network throughput and Mean Opinion Score (MOS). The Mean Opinion Score Listening Quality (MOS-LQ) was used to measure the listening quality. The results were captured at intervals of 5 packet injections.

The algorithm was implemented using an IEEE 802.11s model developed using a network simulation platform ns-3 version 3.36.1. The experiment was run using a computer with Kali Linux version 2022.2 with kernel version 5.18, using 8 GB of RAM and a solid-state drive (SSD) with 100 GB free space.

B. Test results

1) Packet loss

The experiments results showed that packet losses were low with little increments up to about 20 packets at a high of 2.9% with the TP1 scheme as illustrated in Fig 2. With more packets being sent out, the loss percentage grew.

Fig. 4. Packet loss

The losses start to rise more quickly after 60 packets and eventually reach a peak of 15.9%. (TP1). Such packet losses are a sign of the network being congested from the high volume of traffic. The amount of packet loss differed significantly when ROHC was applied to the packets (TP2). The discrepancy got more pronounced when more packets were injected. The initial packet loss between TP2 and TP4 was just about 0.2%, but it visibly rises to about 1.6% after roughly 55 packets. The gap persists because the TP3 packets came in at slightly under 14%, or 1.98% less than the TP1 packets, in contrast to the TP2 and TP4 schemes, which come in much lower at about 3.5% and 2.1%. As evidenced by the TP4 schemes, the differences illustrate that VoIP packet aggregation and compression significantly improves performance.

2) Jitter

The simulation results demonstrated that TP1 exhibited the most jitter, as seen in Fig 3. For the first 25 packet injections, the jitter for TP1 and TP3 packets was practically identical, with the difference being as small as 0.3587 milliseconds (ms). However, as more traffic was introduced, the jitter also increased, especially for the TP1 scheme, as 30 packets at the jitter grew to 0.8652ms and eventually reached 1.2853ms. For the first 20 injected packets, the TP2, TP3, and TP4 had moderate jitter in contrast to the substantial jitter for the TP1. The TP2 packets then increased to 0.6788ms, increased gradually to 0.7625ms, surged at 0.8125ms, increased by tiny bits, and ultimately reached 1.0378ms. The results of the experiments showed that the TP4 packets consistently had the lowest jitter. The findings reveal a slight variation in jitter depending on where traffic is ingested first. This is due to the fact that the aggregation and compression processes used by the TP2 and TP4 aggregation techniques will create packet delays.

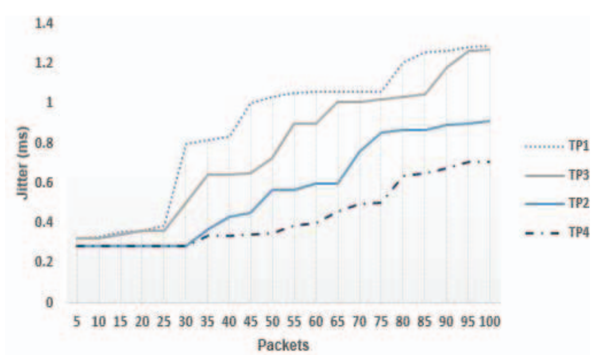
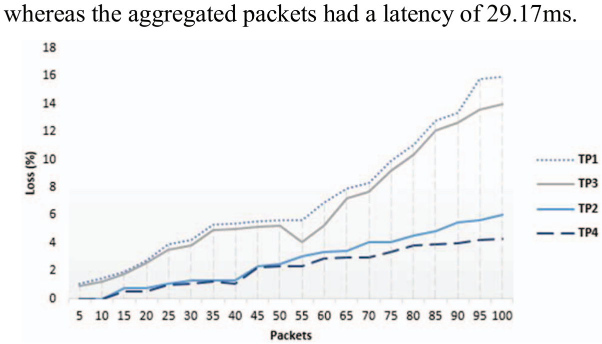


Fig. 5. Results for jitter

3) Latency

End-to-end latency for each category of aggregation is shown. From the experiment, the total delay for the 100 packets utilised in the experiment may be estimated by multiplying the frame transmission delay by the quantity of frames transmitted. The unaggregated packets had an average latency of 18.43ms, whereas the aggregated packets had a latency of 29.17ms.



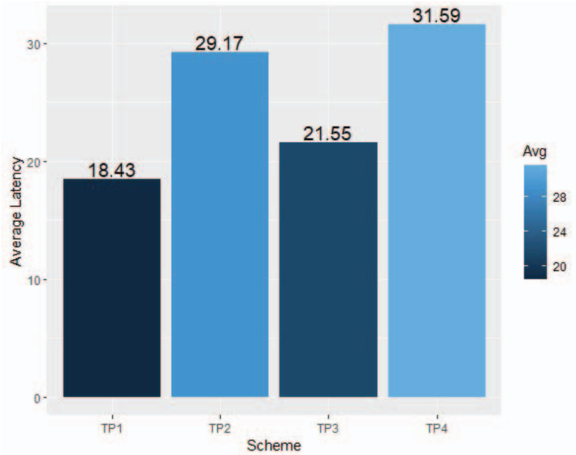


Fig. 6. Results for latency

End-to-end latency for each aggregation category is displayed. By multiplying the frame transmission delay by the number of frames transferred, it is possible to estimate the overall delay for the 100 packets used in the experiment. The aggregated packets had a latency of 29.17ms on average compared to 18.43ms for the unaggregated packets. The packets that were compressed and aggregated had the highest delay, at 31.59ms, compared to the average latency of 21.55ms for the TP3 packets. According to the algorithm, the TP4 scheme's higher latency than the other schemes is brought on by procedures that happen at the origin node.

The algorithm must determine whether any further packets require aggregation, compression, or both. The aggregated packets' average delay is increased as a result.

4) *Network throughput*

The results of the throughput test are exemplified in Fig 7.

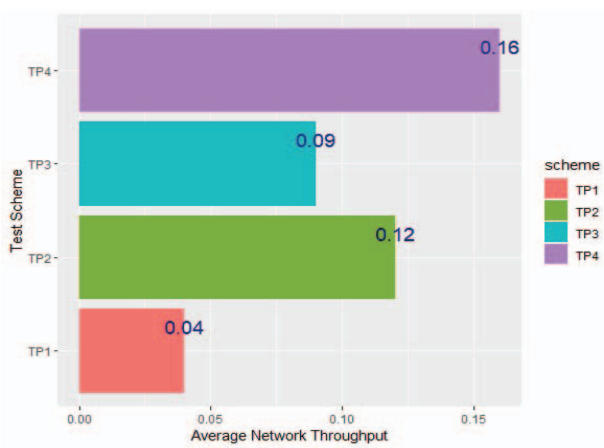


Fig. 7. Network Throughput

The findings demonstrate that the two aggregation-based systems, TP2 and TP4, had a greater throughput than the alternative methods. This shows that when packets are compressed and aggregated, more traffic is transferred. The output was measured in megabytes per second (Mbps). The output was measured in megabytes per second (Mbps). The results show that very high throughput was realised in scheme TP4 than all the other schemes. The average throughput for this scheme was approximately 0.16Mbps. The average throughput for TP1 was approximately 0.04Mbps. The TP2 scheme had an average throughput of 0.12Mbps and the last scheme TP3 had an average throughput of 0.09Mbps.

TABLE II. LATENCY VS THROUGHPUT

Scheme	*Throughput (Mbps)*	*Latency(ms)*
TP1	0.04	18.43
TP2	0.12	29.17
TP3	0.09	21.55
TP4	0.16	31.59

A comparison of the latency and throughput averages was done to get a better understanding of the performance of the schemes. The results are shown in Table II

5) *Mean Opinion Score*

The outcomes for the MOS calculation for all the schemes is shown in Table 5.3. 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 has 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.

TABLE III. MEAN OPINION SCORES

Scheme	*MOS Outcome*
TP1	4.534
TP2	4.489
TP3	4.109
TP4	4.103

C. *Conclusions and Recommendations*

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

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 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 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

D. Future Work

The study looked at the improvement of VoIP over WMN with a high performance standard like 802.11ac. However, it did not cover the aspects of security. Future studies can look into the security implications of utilising both aggregation and compression at the same time in VoIP over WMNs.

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