An Algorithm to Optimize Concurrent VoIP Calls Across Wireless Mesh Networks

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Abstract*—This study focused on improving the performance of Voice over Internet Protocol (VoIP) over Wireless Mesh Networks (WMN) in an 802.11ac scenario simultaneously using packet compression and packet aggregation. The study employed design science research methodology and developed an algorithm that aims to increase the number of concurrent calls while reducing the impact of jitter, delay, and packet drop. The study is the first of its kind to come up with a combined packet compression and packet aggregation technique for Voice over Internet Protocol Wireless Mesh Networks (VoWMN) in an 802.11ac setting. During simulation tests carried out using ns-3, the algorithm outperformed existing schemes in terms of average delay, jitter, packet loss, and network throughput. The algorithm performed better by 1.4% over compressed packets and by 27.22% over plain VoIP packets in terms of packet loss, and obtained the highest network throughput of 0.16Mbps which was 56.25% more than compressed packets which had the next best throughput. These findings suggest that the proposed algorithm can significantly enhance the performance of VoIP over WMN in an 802.11ac scenario. However, this study did not cover aspects related to security implications when utilizing both aggregation and compression at the same time in VoIP over WMNs. Additionally, it only focused on the 802.11ac scenario. Further research could explore the effectiveness of the proposed algorithm in other scenarios and protocols, and identify the optimal settings for the algorithm. Additional research could explore the use of other techniques, such as error correction and QoS mechanisms, to improve the performance of VoIP over WMN.

***Keywords*—Voice over Internet Protocol (VoIP), Voice over Internet Protocol Wireless Mesh Networks (VoWMN), packet aggregation, packet compression**

I. Introduction

Voice over IP (VoIP) technology has become increasingly popular as a means of communication especially when coupled with Wireless Mesh Networks (WMN) which is one of the emerging types of networks technology that are highly available and compatible with VoIP, yet at the same make it easy to deploy VoIP where delivery of voice communication is difficult [1–4]. However, this technology, VoIP over WMN (VoWMN), still faces challenges related to voice codecs, jitter, delay, and packet loss, and these challenges can impact the quality of calls, particularly since WMN has a less robust network infrastructure in comparison to physically wired networks. As a result, a more robust approach is required to minimize these issues and improve the performance of VoIP over WMN [5–7]. This study concentrates on the 802.11ac standard, which boasts a large bandwidth capacity of up to 7GB throughput and operates on the 5GHz frequency range which makes it an optimal technology for use in WMN. Therefore, this research aims to develop an algorithm that combines compression and aggregation techniques to enhance the performance of VoIP over WMN in an 802.11ac setting. The proposed approach endeavors to improve the number of simultaneous conversations while minimizing the impact of delay, jitter, and packet drop [8]. This, in turn, will improve the overall quality of VoIP calls over WMN. The study seeks to fill the gap created by prior studies that have largely concentrated on either packet aggregation or packet compression, with only a few incorporating compression and aggregation techniques and none of the studies have combined packet compression and packet aggregation as a means of improving call quality in an 802.11ac setting.

This essay is structured as follows: The related work is briefly summarized in Section II, the technique employed in this study is described in Section III, the experimental results and analysis are presented in Section IV, and the paper is concluded with recommendations for further research in Section V.

II. Related Work

A. Overview of VoIP and VoWMN

VoIP systems enable real-time transmission of voice signals in the form of data packets across Internet Protocol (IP) networks, in contrast to PSTN, which offers dedicated end-to-end circuit connections for the duration of each conversation. Due to its utility in communication, VoIP has become a very popular technology as pointed out by the authors of [7–9], spanning a host of popular VoIP technologies or applications like Skype, Viber, WhatsApp, and a variety of other technologies with VoIP capacity. According to Montazerolghaem [2], VoIP technology has become more widely used since it provides the highest degree of service quality, is reasonably priced, and is more

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reliable. Even though 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 realized. 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 the quality of service according to studies by the authors of [8, 10–12].

Voice over Internet Protocol Wireless Mesh Networks (VoWMN) are described by [13] as a very attractive way to extend the network coverage into the dead zone in cases where the wired network is not easy to install. WMN are described as a multi-hop wireless network made up of communication nodes following a mesh topology according to [14]. Additionally, authors of [15, 16] point out that wireless mesh networks help to enable individuals to be continuously online thereby solving problems associated with wired networks. As the author of [17] put it, the combination of Wireless Mesh Networks (WMN) with VoIP is an attractive solution for enterprise infrastructures. According to the authors of [13, 18], a mesh network has a comparative advantage over wired Local Area Networks (LANs) because of ease of deployment ease of expansion, better coverage, robustness to node failure, and reduced cost of maintenance.

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 as pointed out by Kim and Kong *et al*. [13]. Fig. 1 presents a typical VoWMN network architecture.

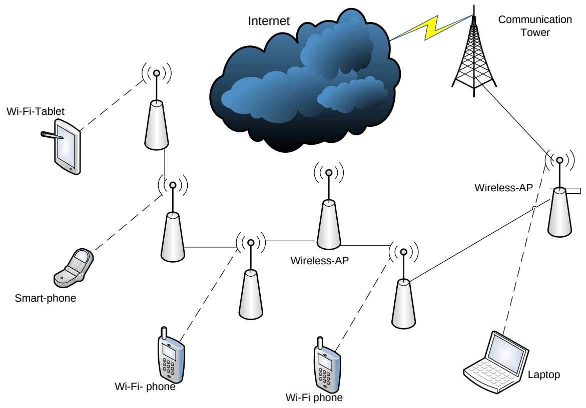

Figure 1. Typical VoWMN architecture.

B. *WMN and VoIP Routing Protocols*

Meeran and Annus *et al.* [19] established that while node mobility can increase packet loss, delay, and jitter, VoIP implementations in WMNs are more efficient in cases where there is no background traffic. Therefore, recommends that VoIP implementations in wireless mesh networks should have mobile nodes which support background traffic thereby increasing the quality-of-service standards and assurance of packet data delivery [20].

Similarly, studies by Meeran and Annus *et al.* [19] reveal that VoIP quality of service in WMN can be improved by identifying the integration choices and inclusion of supportive mesh nodes. The research in which they used a network simulator and experiments conducted on three main scenarios with mesh nodes in no-mobility, partial mobility, and full mobility deployments. 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 according to [20].

Kirsanova and Radchenko *et al.* [21] proposed the use of cloud-based systems as an alternative solution to enabling cost-quality optimization of VoIP systems with three configurable parameters for model bases on bin packing namely utilization threshold, rental threshold, and Prediction Interval (PI) as mechanisms to cope with different objective preferences, workloads, and cloud properties. As the authors propose, this in turn can be dynamically adapted to environmental changes.

Furthermore, Chethan and Basavaraju *et al.* [22] 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 optimum level because it is closely monitored and maintained.

Amiri and Prakash *et al.* [23] proposed 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 network's life. 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. Another proposal was made by Darabkh and El-Yabroudi *et al.* [24], called the 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. BPA-CRP technique assigns four different broadcast ranges for each sensor. A compressed fuzzy logic-based multi-criteria Ad hoc On-demand Distance Vector (AODV) whereby routing decisions are dependent on the number of relays, distance factor, direction angle, and vehicle speed variance is proposed by Fahad and Ali *et al.* [25]. An earlier study by Jain and Chawla *et al.* [26] presented nearly similar recommendations. A QoS-aware routing protocol with an adaptive feedback scheme for video streaming for mobile networks is advanced by Castellanos and Guerri *et al.* [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. Alghamdi [28] is of the opinion that a Load-Balancing Ad hoc On-demand Multipath Distance Vector (LBAOMDV) routing protocol is more appropriate. According to Alghamdi [28], "LBAOMDV regulates the fair usage of both node energy and available bandwidth by exploiting the availability of multiple paths for data transfer".

Bhattacharjee and Bandyopadhyay [29] recommended an energy-efficient routing algorithm that strives to strike a balance between data traffic among the nodes and network lifetime using the 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

Past studies suggest that VoWMN is an evolving phenomenon. 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 as Meeran and Annus *et al.* [19], Shahdad and Sabahath *et al.* [30] point out. Meeran and Annus *et al.* [19] point out that WMNs are an emerging network system that is suitable for various applications including harsh environments such as military fields and emergency relief situations.

Soloviev and Solovieva *et al.* [31] 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. Abualhaj and Shambour *et al.* [9], Parvin [32] state that VoIP over WMN suffers from inefficient bandwidth use because attaching a 40-byte RTP/UDP/IP header to a small VoIP payload and 841µs causes delay and overhead of each packet in WMNs. The authors then suggest VoIP packet multiplexing as the most appropriate solution to the problem. Shahdad and Sabahath *et al.* [30] identify the robustness, reliability, and speed of WMN as one of the main reasons why they have been adopted to enhance services in various areas like broadband home networks, transport Systems, health and medical sciences, security, and surveillance systems, disasters reporting and emergency.

Packet aggregation which is defined by Zulu [33] as a means of combining small multiple packets together to form a larger packet, is a promising technique to increase WMN's efficiency. Various aggregation algorithms have been proposed for purposes of increasing WMN capacity such as one proposed by Akyurek and Rosing [34]. Zulu [33] identifies two aggregation methods which are hop-to-hop aggregation which involves the de-aggregation of packets at one hop, followed by aggregation at the next hop. This implies that after packets are de-aggregated, they are aggregated when they are ready for transmission. The other aggregation method is end-to-end aggregation in which packets are only aggregated on the sending node and aggregated at the receiving node. 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 till they reach the destination where de-aggregation takes place [33].

Marwah and Singh [35] however opines that packet aggregation is justified up to a certain extent only as after that most of the users get dissatisfied. To counter that, Akyurek and Rosing [34] proposed an optimal packet aggregation algorithm with three levels of flexibility namely: per application data stream; per packet; and, under different network conditions.

Neves [36] provides an alternative proposal for increasing WMN capacity. The authors suggest that optimal classification of voice packets is necessary in order to enhance 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 and Rosing [34], the proposed method is based on a dynamic programming optimization algorithm that finds the optimal subset of m high-priority voice segments in each utterance of size $n > m$.

A study by Olariu and Fitzpatrick *et al.* [5] propose 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 queuing delay that has been accumulated across multiple hops within the WMN. Using the DPPM, VoIP packets that have been in the queue for longer are given higher priority over less delayed VoIP packets. The proposed DPPM is also used to ensure voice call quality and capacity are enhanced in combination with Wi-Fi frame aggregation according to [5].

An earlier study by Azevêdo and Caetano *et al*. [37] made almost similar suggestions to the study by Olariu and Fitzpatrick *et al.* [5]. They propose a packet aggregation technique called 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 and Liu *et al*. [38] 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 the transmission volume and communication overhead. Further, each packet is divided into forwarding packets and aggregating packets, with the forwarding packets being used to complete the service aggregation of aggregating packets to minimize the transmission volume and communication overhead without additional latency. The study showed that the algorithm reduces response delays are reduced by 31.33% to 51.41% which in turn leads to a better quality of experience among users according to Huang and Liu *et al*. [38].

According to studies by Dely [39], Castro and Dely *et al*. [40] adaptive aggregation supports twice as many VoIP calls in comparison to a fixed maximum packet size aggregation. Thus, the two studies suggest aggregating small packets for VoIP traffic. Asif and Shafiq *et al*. [41] suggest 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 of the study by Asif and Shafiq *et al*. [41] shows that AADM outperforms existing static approaches in terms of packet loss, throughput, and delay.

Packet compressions in [33] showed 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 Brocke and Hevner *et al.* [42], the mobile SCTP-Concurrent Multipath Transfer (mSCTP-CMT) technique could be an alternative model. The authors found that mSCTP- CMT can improve the overall throughput in homogenous networks. They therefore propose an mSCTP-based Bandwidth Aggregation, (msctp-BA) technique that enables efficient use of the resources of heterogeneous network technologies like WLAN and UMTS. Reports from their simulation show that the mSCTP technique increases user experience and quality of service because it was better by 23% for VoIP and file download applications.

III. METHODOLOGY

The study employed the design science research methodology. Brocke *et al.* [43] defines 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 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. Structure of a VoIP Packet

A VoIP packet has two major divisions, the packet header, and the payload which carries the encoded voice data. The header is made up of the Internet Protocol (IP) header, then the User Datagram Protocol (UDP) header, and then the Real Time Protocol (RTP) header. Those three headers constitute the header of a VoIP packet [9].

The payload or the data then comes after the headers. The structure of a typical VoIP packet is shown in Fig. 2. The header is composed of three components, the IP header is 20 bytes, the UDP header is 8-bytes, and the RTP header with a minimum of 12 bytes. All in all, the header is 40 bytes minimum. In a VoIP packet, the payload size depends on the codec being used.

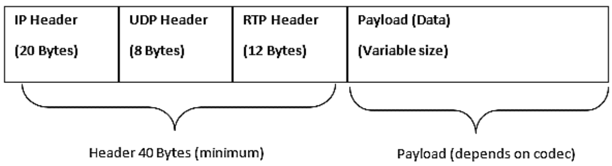

Figure 2. Structure of a VoIP Packet.

B. VoWMN System Architecture

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 each other [32]. Devices in the network include nodes, clients, routers, gateways, and so on. There are three types of nodes in a WMN which are clients, routers, and gateways. The WMN clients are the end-user devices such as laptops, smartphones that can access the network. WMN clients can then use applications like browsing the World Wide Web, VoIP, playing games, location detection and various other tasks. WMN clients are assumed to be mobile with limited power and oft times have routing capability, and may or may not be always connected to the network.

The second components are WMN routers are in the network to route the network traffic. They cannot terminate nor originate the traffic. The routers have limitations in mobility and they have reliable characteristics. Transmission power consumption in mesh routers is low, for multi-hop communications strategy. 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. Since the gateways in WMNs have multiple interfaces to connect to both wired and wireless networks, they are expensive. Therefore, there are a few numbers of WMN gateways in the network. Moreover, their placement has a significant impact on the performance of the network

C. 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 Sun and Dong *et al.* [44] in a typical VoIP packet, the payload accounts for only 33% of the total size of the packet with the rest going to the header. To save bandwidth, VoIP applications mostly utilize header compression since this takes up a bigger chunk of the packet.

Header compression makes use of the fact that most header fields only change a 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, like RTP hear compression and Robust Header Compression (ROHC). For this study, ROHC was selected because it is also applicable over wireless links. ROHC described in RFC 5225, can reduce the overhead to one byte per packet. Fig. 3 shows how ROHC compresses all the separate headers into one header, which is known as the ROHC Header. During the transmission of a VoIP packet, only the first packets contain redundant information. As shown in Fig. 3, compressing all the various static information that will be needed is compressed into one header, the ROHC Header which will then be transmitted across the network.

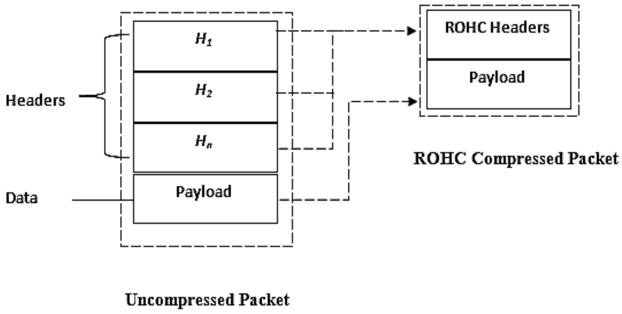

Figure 3. ROHC compression.

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.

D. Packet Aggregation

Packet aggregation means 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. Fundamentally, the packet aggregation approach works by collecting packets at a common node which is also called the aggregation target, then merging the packets into one big packet, and then forwarding this new packet to the destination node (de-aggregation target) where the packets are disassembled in a process known as de-aggregation. At the destination, the packets are de-aggregated, and the receiver gets the correct packets. Packet collection is done in the MAC layer which is the ideal place to do this. All essential data about a packet, including the IP and MAC addresses of the next node or destination node, are accessible at this point at the MAC layer making it the best place for collection of packets, and new IP and MAC headers are created. The old MAC header is destroyed, but the old IP header is kept. 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 disaggregation module as illustrated in a high-level description of the concept in Fig. 4. Packets $P(0\text{-}n)$ are aggregated into one big packet Pa and then transmitted across a network.

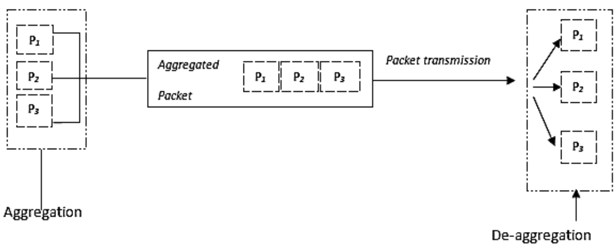

Figure 4. High-level aggregation.

Fig. 5 represents the original packet structure before aggregation, as well as the combined packets after aggregation.

In Fig. 5, there are three packets, each with its own header and payload. Through aggregation, the headers are replaced with the aggregated headers, and the payload become one. Eq. (1) shows the basic aggregation where P_i to P_n are aggregated to form a new packet.

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

As a way of identification of the various packets that make up the new aggregated packet P_a and their aggregation targets (destination nodes), 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 is the end of the aggregation tunnel for end-to-end aggregation. Packet aggregation, therefore, basically means that packets going to the same next hop or destination, are concatenated, and prepended with 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.

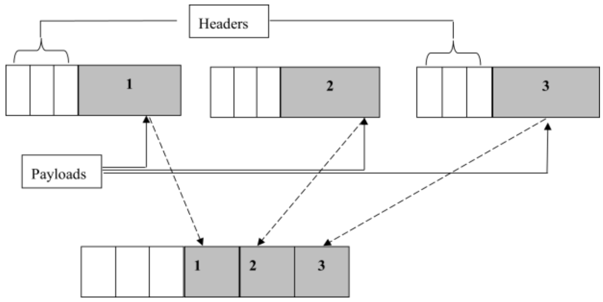

Figure 5. Simplified packet aggregation.

Fig. 6 shows a detailed illustration of packet aggregation. The diagram, from [45] shows the time saved by aggregation and also the data savings. Initially, three different packets are shown, each with its own header DIFS, Back-Off (BO) algorithm, and payload. On the receiver's end, each packet has its own SIFS and gets an acknowledgement.

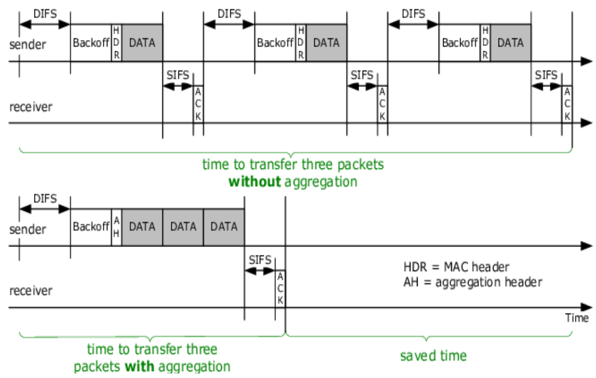

Figure 6. Detailed packet aggregation.

Through aggregation header redundancy is removed, and only one set of BO, DIFS, SIFS, and acknowledgement is kept. Instead of the MAC header for each packet, an aggregation header is introduced. It has the information on each packet's origin. Therefore, as shown in Fig. 6, aggregation saves time and other overheads associated with sending multiple packets across the network.

E. Determination of Optimum Packet Size

The concept of aggregation also includes the need for the determination of the maximum size of the packet that will be transmitted. According to the size of the packet that is 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 MTU for a VoIP packet is 1460 bytes, even though majority of the broadband routers are set to an MTU default of 1454 bytes. Therefore, as illustrated in Eq. (2), with m as the limit MTU, the limit of the packet x can be identified as

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

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

F. Determination of Maximum Waiting Time

The time to wait (or aggregation delay) will be implemented using the Holding Time Packet Aggregation [38]. Let A_{max} be the maximum allowed time for a packet P to traverse the path $P(s \rightarrow d)$, where s is the source node and d is the destination node. Node n_r is a relay node on the path $P(s \rightarrow d)$. 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 Eq. (3).

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

where |P (r, d)| is the number of hops in the $P(r \rightarrow d)$ path, that is, the path from the relay node nr to the destination node [38].

IV. Proposed Algorithm

The proposed algorithm is comprised of two phases which are done in sequence. The first phase is the compression of the VoIP packet and which will be followed by the aggregation of the VoIP packets. The proposed algorithm will start by compressing packets with the same target node at the point of origin using ROHC.

Thereafter, each compressed 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 in its way to the destination node. At the destination node the packets are de-aggregated and reassembled and then decompressed. Fig. 7 illustrates in principle how the algorithm will work.

Fig. 7 provides an overview of how the proposed algorithm will work in a network using end-to-end aggregation. Algorithm 1 shows how the proposed algorithm will function. The algorithm is explained in the following passages.

Given 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 done at the nodes of origin. At node $n0$, there are three packets P_1, P_2, and P_3 and node $n1$ has one packet P_n originating from it. Each packet is compressed at the source node regardless of whether it will be aggregated or not. At the source node, packets with the same destination are then aggregated. 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, then the packets are aggregated and a new aggregated packet P_a composed of P_1, P_2, and P_3 is created.

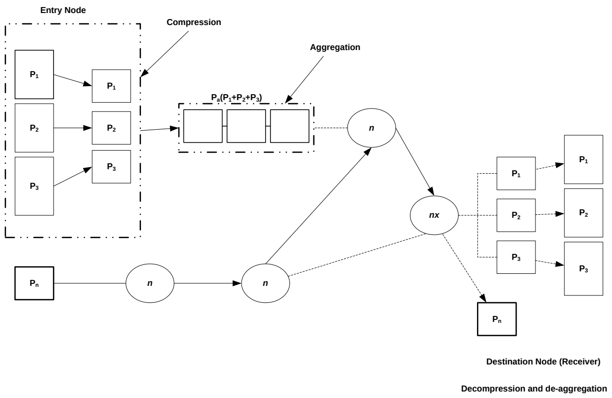

Figure 7. The proposed algorithm in use in a network.

Algorithm 1. Adaptive VOIP Optimization Algorithm

Main Procedure
//Initialize Variables
get cP //current VoIP packet
get 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

As shown in Fig. 7 and Algorithm 1, other packets will be added to P_a, given that P_a has not reached the maximum MTU or the maximum holding time has not elapsed. That is, if no other packet arrives within the set timeframe, 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 sending 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, then 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 may traverse the network using a different path.

At the destination node, all the packets are de-aggregated first to get the original packets P_1, P_2, and P_3 from P_a. The packets are then decompressed to provide the payloads. Since VoIP traffic is sensitive to delay, a packet that does not get aggregated within the maximum delay time, like P_n, is immediately released from the waiting queue and transmitted without being aggregated.

V. Experimental Setup

A. Simulation Parameters

The algorithm was implemented using a model developed using a network simulation platform ns-3 version 3.36.1. The experiment parameters are shown in Table I.

The experiments consisted of 100 VoIP packets of differing sizes (from 20 bytes to 80 bytes), being injected into the network. The experiments used the iLBC VoIP codec. The packets originated from either source node or 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.

TABLE I. Simulation Parameters

No.	Item	Description
1	Simulation Tool	ns-3 version 3.36.1
2	Network Protocol	IEEE 802.11s
3	Number of packets	100
4	VoIP Protocol	iLBC
5	Initialization	Random
6	Channel Type	Wireless Channel
7	Type of Network Interface	Wireless
8	Computer RAM	16 GB
9	Packet size	20 bytes–80 bytes
10	Number of nodes	53
11	Routing Protocol	Hybrid Wireless Mesh Protocol (HWMP)
12	Simulation Area	VoIP packet performance

B. Performance Parameters

This section discusses the performance parameters that were 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 we compared are as follows:

1) The first metric to be measured is packet loss. A packet drop occurs when packets are lost throughout the routing process between the sender and the recipient for a variety of reasons.
2) The second metric to be measured was jitter. Jitter is the variation of packet delay that is caused by queuing lengths, traffic, and the use of different routes throughout the network.
3) The third metric that was 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.
4) The fourth metric to be measured was network throughput. This is the amount of data transmitted over a network in a given period of time.
5) The final metric to be measured was the Mean Opinion Score (MOS). The MOS measures the 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 that is based on several factors like signal-to-noise ratio.

C. Test Cases

The following describes the different packet categories that were used in the tests. A total of four categories were created, including the category using the Adaptive VoIP Optimization Algorithm (AVOA).

1) The first category that was used in the tests was coded *PP* for Plain Packets. These are packets that are sent as they would without any compression or aggregation save for that which is done by the VoIP codec.
2) The second scheme was named *PC* for compressed packets. These packets are compressed using ROHC only without any aggregation.
3) The third scheme was named *PA*. These packets are aggregated only using end-to-end aggregation. These packets did not have compression done on them.
4) Then the final category of packets are those that were named *PCA* for compressed and aggregated packets, which used the AVOA.

VI. Discussion

A. Packet Loss Outcome

The experimental results showed that packet losses were low with little increments up to about 20 packets at a high of 2.9% with the PP category as illustrated in Fig. 8. With more packets being sent out, the loss percentage grew. The losses start to rise more quickly after 60 packets and eventually reach a peak of 15.9% (PP category). Such packet losses are a sign of the network being congested by the high volume of traffic. The amount of packet loss differed significantly when ROHC was applied to the Packets (PC). The discrepancy got more pronounced when more packets were injected.

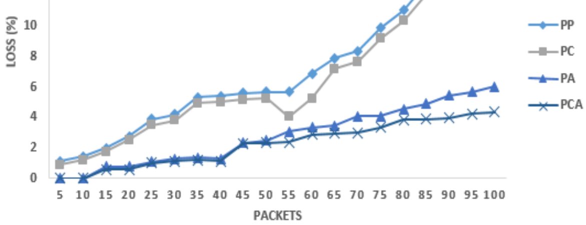

Figure 8. Results for packet loss.

The initial packet loss between PC and PCA was just about 0.2%, but it visibly rises to about 1.6% after roughly 55 packets. The gap persists because the PC packets came in at slightly under 14%, or 1.98% less than the plain packets, in contrast to the PA and PCA schemes, which come in much lower at about 3.5% and 2.1%. As evidenced by the PCA category packets, the differences illustrate that VoIP packet aggregation and compression significantly improved the performance. There is a 27.22% improvement in packet loss when comparing the packets from the PCA category and those from the PP category.

B. *Jitter*

The simulation results demonstrated that PP exhibited the most jitter, as seen in Fig. 9. For the first 25 packet injections, the jitter for PP and PC packets were 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 PP category, as at 30 packets the jitter grew to 0.8652 ms and eventually reached 1.2853ms. For the first 20 injected packets, the PA, PC, and PCA had moderate jitter in contrast to the substantial jitter for the PP category. The PA packets then increased to 0.6788 ms, increased gradually to 0.7625 ms, surged at 0.8125 ms, increased by tiny bits, and ultimately reached 1.0378 ms. The results of the experiments showed that the PCA packets consistently had the lowest jitter. The findings reveal a slight variation in jitter depending on where traffic is ingested first. This is because the aggregation and compression processes used by the PA and PCA aggregation techniques will create packet delays.

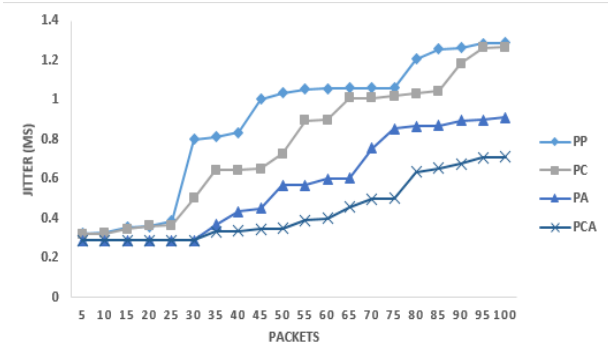

Figure 9. Results for jitter.

C. *Latency*

End-to-end latency for each category of aggregation is shown in Fig. 10. 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 number of frames transmitted. The unaggregated packets had an average latency of 18.43 ms, whereas the aggregated packets had a latency of 29.17 ms.

The PCA category had the highest jitter. This is because the algorithm must determine whether any further packets require aggregation, compression, or both. The aggregated packets' average delay is increased as a result.

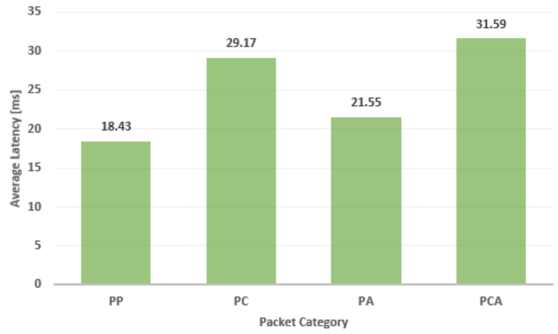

Figure 10. Average latency.

D. *Network Throughput*

The results of the throughput test are exemplified in Fig. 11. The findings demonstrate that the two aggregation-based systems, PA and PCA, 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 a very high throughput was achieved in scheme PCA than all the other schemes. The average throughput for this scheme was approximately 0.16 Mbps. The average throughput for PP was approximately 0.04 Mbps. The PC scheme had an average throughput of 0.12 Mbps and the last scheme PA had an average throughput of 0.09 Mbps.

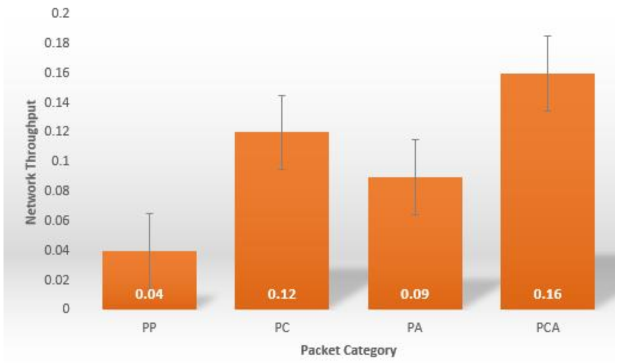

Figure 11. Network throughput.

The results in Fig. 11 show that the PCA category had a network throughput of four times the PP category. According to the results of the experiments, the use of the AVOA algorithm, improved network throughput by as much as four times in comparison to plain packets. 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.

TABLE II. LATENCY VS THROUGHPUT

Packet Category	Throughput	Latency
PP	0.04	18.43
PC	0.12	29.17
PA	0.09	21.55
PCA	0.16	31.59

E. Mean Opinion Score

The outcomes of the MOS calculation for all the schemes are shown in Table III. The findings show that the highest MOS is achieved with the PP scheme at around 4.5 followed by the PC category with 4.489 and the PA category has 4.109. The PCA category 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
PP	4.534
PC	4.489
PA	4.109
PCA	4.103

VII. CONCLUSIONS AND FUTURE WORK

The study found that the proposed algorithm improved the performance of VoIP over WMN in an 802.11ac scenario by increasing the number of concurrent calls while reducing the impact of jitter, delay, and packet drop. The algorithm was compared against three existing schemes: 1) end-to-end aggregated traffic, 2) ROHC compressed-only traffic, and 3) plain traffic without aggregation or compression. The proposed algorithm outperformed existing schemes in terms of average delay, jitter, and packet loss. The findings suggest that the proposed algorithm can significantly enhance the performance of VoIP over WMN in an 802.11ac scenario. The results showed that the proposed algorithm outperformed existing schemes in terms of average delay, jitter, and packet loss rate. These findings suggest that the proposed algorithm can significantly enhance the performance of VoIP over WMN in an 802.11ac scenario.

A. Academic Implications

The findings of this study have important implications for researchers, network engineers, and service providers who are interested in improving the performance of VoIP over WMN. The Adaptive VoIP Optimization Algorithm can be used as a valuable contribution to this field of research and can be incorporated into existing schemes to enhance their performance. Therefore, the study contributes to the existing body of knowledge on wireless mesh networks and VoIP, and could potentially lead to further research and development in this area.

B. Limitations of Study

One potential limitation of this study is that it only focused on the 802.11ac scenario. More research is needed to determine whether the proposed algorithm would be effective in other scenarios or under different network conditions. Additionally, the study focuses on the performance of VoIP across WMN without considering the aspect of the security of the voice packets themselves.

C. Directions for Future Studies

Further research could explore the effectiveness of the proposed algorithm in other scenarios and protocols, and identify the optimal settings for the algorithm. Additional research could explore the use of other techniques, such as error correction and QoS mechanisms, to improve the performance of VoIP over WMN. Further studies could also explore the feasibility and practicality of implementing the proposed algorithm in real-world networks. Finally, researchers could investigate other techniques for improving the performance of VoIP over wireless mesh networks, such as using multiple access points or optimizing the routing protocol.

CONFLICT OF INTEREST

The authors declare no conflict of interest.

AUTHOR CONTRIBUTIONS

Olusola Olorunnisola carried out the research and wrote the paper; Topside E. Mathonsi and D. Du Plessis analyzed data; all the authors approved the final version of the paper.

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