A Dynamic Packet Aggregation Scheme for VoIP in Wireless Mesh Networks

J. Okech, Y. Hamam, *Senior Member IEEE*, A. Kurien, *Member IEEE*, T. Olwal, M. Odhiambo

Abstract— Wireless mesh networks (*WMNs*) based on IEEE 802.11 standard is a new trend in networking combining performance, simplicity and economics suitable for backhaul deployments. However, service providers and carriers understand the cost of making large scale adoption of infant technologies. If *WMNs* are to compete with existing broadband networks, then the technology must evolve from low cost consumer deployment to carrier-grade structures with improved performance and robustness. *WMNs* have brought unique challenges because its broader coverage calls for accommodation of increased number of clients with varied service demands. For *WMNs* backhaul with voice over Internet protocol (*VoIP*) clients, the stringent quality of service (*QoS*) requirements present additional challenge. This paper proposes a dynamic packet aggregation algorithm that adjusts the size of aggregation packet to improve *VoIP QoS* in *WMNs*. Simulation results show that the proposed algorithm reduces end-to-end delay, jitter and packet loss for *VoIP* packets in *WMNs*.

Index Terms—Aggregation, QoS, *WMNs*, *VoIP*.

I. INTRODUCTION

Voice over IP (*VoIP*) network systems are becoming suitable alternative to the traditional public switched telephone networks (*PSTNs*) in both corporate and residential areas by reducing the cost for networking, management and support. The cost saving feature is attributed to the use of existing data infrastructures and is the main factor fuelling this steady growth.

With the upsurge in the popularity of IEEE 802.11 based networks in homes and offices, wireless *VoIP* has become a more attractive adventure providing the caller with more expediency. For example, wireless local area networks (*WLANs*) make it easier for users to access telephone services anywhere anytime through portable handsets. However, existing internet connectivity is provided via wired access point (AP) that is costly and inflexible to deploy for wider coverage.

The *WMNs* use mesh routers to extend access point (AP) coverage to areas where wired networks are not easy to install or uneconomical to set up. Thus, *WMNs* provide a viable alternative for creating an enterprise-scale or community-scale wireless backbone. Such a structure not only supports multiple users but also drives these users from using existing fixed phones to wireless *VoIP* phones.

Nonetheless, in *WMNs*, the number of supported calls for two way conversation drops as the number of *VoIP* sessions increases. For IEEE 802.11 based *WMNs*, the main challenge in providing higher packet transfer ratio lies on management of the medium access control (*MAC)* protocol overhead [1]. This overhead is attached to every packet on transmission and hence consume significant portion of network bandwidth. Thus, the dismal performance due to this overhead magnifies for small packets such as *VoIP*. In this work, a packet aggregation mechanism that dynamically determines and adopts optimal aggregation packet size based on link quality characteristics is proposed to improve *VoIP QoS* over *WMNs*.

The rest of this paper is organized as follows: In Section II, related work is discussed. Section III details out how protocol overhead impacts on *VoIP* call capacity for *VoIP* over *WMNs*. Aggregation algorithms are analysed in Section IV. In Section V, simulation results are presented and discussed and Section VI concludes the work.

II. RELATED WORK

Dealing with transmission of small packet size traffics in IEEE 802.11 based networks has been a long standing issue. Authors such as Hole and Tobagi [2] found that each *AP* can only support a few *VoIP* flows due to the large overhead of IEEE 802.11 *MAC* in processing small packets. Studies conducted to understand the capacity of *WMNs* in [1] show that the throughput of each node decreases at order $O(1/n)$, where n is the number of hops.

Several approaches have been proposed to solve this anomaly both for single and multi-hop networks. However, for this work, only the literature related to the proposed methodology will be discussed. The use of packet aggregation to improve performance of *VoIP* application on the network is presented in [3], [4], [5] and [6]. The basic decision for an aggregation algorithm in *WMNs* is the placement of de-aggregation capability. This choice defines the applied packet aggregation mechanism. There are two basic approaches to packet aggregation: *end-to-end* aggregation and *hop-by-hop* aggregation. In *end-to-end* aggregation, packet aggregation takes place at the ingress nodes while the egress nodes do the de-aggregation. The *hop-by-hop* aggregation does aggregation at every node

J. M. Okech, A. Kurien and Y. Hamam are with French South African Technical Institute in Electronics, Tshwane University of Technology (TUT) P/ Bag X680, Pretoria – 0001, South Africa (okechjr@gmail.com, {kurienam, hamama}@tut.ac.za).

T. Olwal is with TUT and the Meraka Institute at CSIR (tolwal@csir.co.za).

M. O. Odhiambo is with University of South Africa (ohangmo@unisa.ac.za).

along the source to destination path. Important parameters for implementing packet aggregation are allowable aggregation packet size and the aggregation delay time. These parameters can be implemented as fixed, dynamic or a combination of fixed and dynamic at various parts of the network to yield varied results. Thus, a suitable mix of these basic aggregation parameters can be used to design better aggregation method.

In [3], the use of concatenation mechanism to reduce protocol overhead is proposed. It assumes a network with homogeneous nodes. This assumption presents an inefficient usage of bandwidth. In [4], IP based adaptive packet concatenation algorithm for multi-hop *WLANs* is proposed and simulated. The simulation results reveal that more than double the throughput can be achieved in highly loaded networks but at the expense of increased end-to-end delay. The authors in [5] describe IEEE 802.11 overhead and the importance of packet aggregation in Ad Hoc networks. Two aggregation algorithms are proposed: *forced* algorithm and *adaptive* algorithm. The *forced* algorithm introduces additional delay at every hop from source to destination. The algorithm can result in higher cumulative delay which is not suitable for real-time application. On the other hand, the *adaptive* algorithm proposed in [5] does not usually have sufficiently enough packets to aggregate to provide good bandwidth savings. The authors in [6] investigate the impacts of aggregating multiple small *VoIP* streams in wireless networks. The results of the experiment reveal the existence of relationship between number of *VoIP* calls, output link rate and certain teletraffic metrics. However, the aggregation algorithm used a link rate which is not adjustable to the network situation.

Frame aggregation and optimal frame size adaptation for IEEE 802.11 *WLANs* are presented in [7] and [8]. In [8], a model for calculating the successful transmission probability of a frame of a certain length is proposed. The results of this experiment show that the levels of network contention only has a minor influence on transmission and that *dynamic* aggregation outperforms *fixed frame* aggregation. However, the paper fails to detail out how the frames are delayed. It was developed and verified for single-hop where only self interference is more prominent. These situations do not apply to *WMNs*. In [7], a method to adapt the frame size dynamically to the channel quality and network contention is presented. By intermarrying *end-to-end* and *hop-by-hop* aggregation algorithms, the proposed *accretion* algorithm exploits the advantages of the two while also routing out their shortcomings. The *accretion* algorithm uses forced delay at the ingress to collect packets of the same flow and natural media access delay for intermediate nodes. The paper shows that for higher offered load, the optimum frame size increases up to a dropping point. Thus, it is beneficial to reduce the channel rate and obtain optimal packet size to minimize the interference and contention in a link.

III. VOIP OVER WIRELESS MESH NETWORKS

A. WMNs Architecture

The *WMNs* are built on a mix of fixed backhaul mesh routers and fixed or mobile mesh clients as shown in Fig. 1. The clients can be Wi-Fi enabled *VoIP* handsets, laptops or any other wireless handheld devices that have wireless connections across the *WMNs* to other wired or wireless devices. Communication from these clients go through the local mesh network to other *VoIP* phones via the Internet with the help of gateways or to *PSTN* through local private branch exchange *PBX* [7]. Wired or wireless nodes that extend network coverage are called mesh routers. These routers provide backhaul connectivity at the link level or network layer.

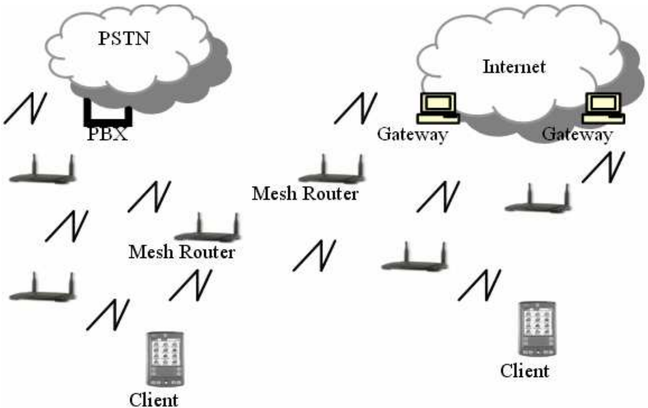


Fig. 1. Voice over WMNs. Communication paths are maintained among mesh routers. Each mesh router has enough interfaces to connect to clients and backhaul. Clients can connect to fixed wireless client, internet or to PSTN through the PBX.

Typical IEEE 802.11 nodes may use Distributed Coordination Function (*DCF*) or Point Coordination Function (*PCF*) MAC access protocols. Although *PCF* offers better support for the *QoS* needs of real-time traffic, it is uncommon and is almost never deployed. This work assumes IEEE 802.11b Wi-Fi standard with *DCF* channel access for all wireless nodes.

B. VoIP Call and the overhead in IEEE 802.11 based WMNs

VoIP systems use codecs to link the digital and analogue interfaces in a communication channel. The codec receives analogue voice, converts it to packets and releases them at a defined rate. There are several vocoders available in the market today such as G.711 [9], G.723 [10] and G.729A [11] each coming with its pros and cons. However, the G.729A is becoming more popular. For simulation purposes this paper models VoIP traffic assuming G.729A parameters.

The G.729A vocoder generates 20 bytes VoIP payload at a rate of 50 packets every second. It therefore means that after adding the 40 bytes *IP/UDP/RTP* header, the minimum channel capacity to support a voice stream in one direction is 24Kbps. This approximates to about 229 *VoIP* calls for 11Mbps channel. However, experimental and analytical results such as in [7] indicate that there is low *VoIP* call capacity. The decrease in capacity can be attributed to larger aggregate time spent by network in sending headers and acknowledgements, waiting for inter-frame separations, and contending for the medium. For example, the 20 bytes *VoIP* payload contribute $14.5\mu s$ at 11Mbps, but *IP/UDP/RTP* header, *MAC* headers and physical headers, trailers, inter-frame periods, back-off and acknowledgements (*ACK*) need a total of $818\mu s$. The contribution of the *VoIP* payload leads to a cumulative transmission time of $832.5\mu s$. This is about 12 *VoIP* calls supported per hop. The number of supported calls is calculated using:

$$(2.\beta.\alpha)^{-1}, \qquad (1)$$

where β is the number of packets generated by vocoder per second and α is the total transmission time for VoIP overheads. These analyses reveal that the per-frame overhead in the IEEE 802.11 standard significantly limits the capacity of *VoIP* over *WMNs*.

Apart from protocol overhead, providing quality *VoIP* services faces additional challenges when exposed to channel noise and interference. Channel noise and interferences increase with increase in number of flows, a feature common with *WMNs*. However, it is imperative to note that packet aggregation can be adopted to improve the performance of *VoIP* over *WMNs*.

IV. PACKET AGGREGATION ALGORITHMS

Aggregation algorithms combine several small packets into one larger packet and forward the larger packet to an aggregation target. The aggregated packet is preceded by a mini-header in which there is an identifier (*ID*) for the session of the flow. The aggregation target uses the *ID* to recover *VoIP* packets out of the combined packet. Fig. 2 summarizes the aggregation process.

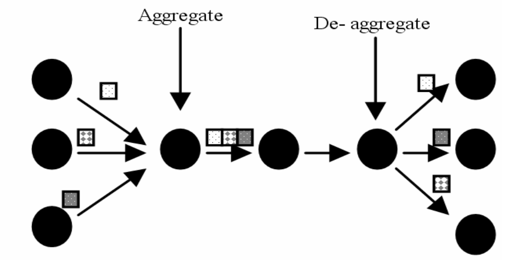
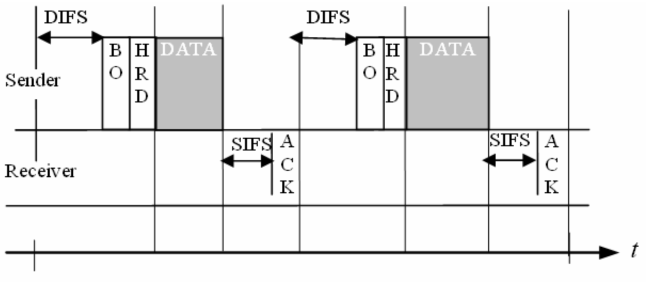


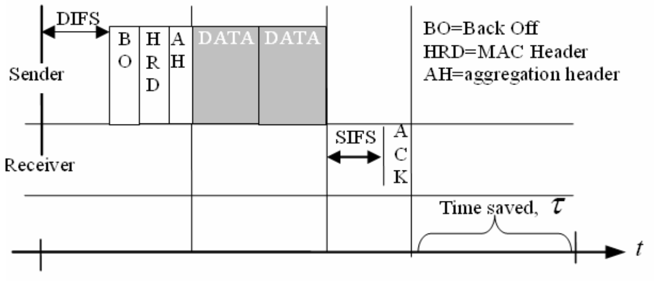
Fig. 2. Packet Aggregation

The process of packet assembly and dis-assembly can be done at the *MAC* or network layer. Packet assembly is done closer to the source of traffic. The recovery process is known as fragmentation or de-aggregation depending on the layer in which aggregation is done.

As illustrated in Fig. 3 (a) and (b), packet aggregation can boost the throughput of IEEE 802.11 based *WMNs*. The figure shows how channel is used during the transfer of two packets with and without aggregation. As has been discussed in Section III A above, the transfer times for one and two *VoIP* packets in IEEE 802.11 network is $832.5\mu s$ and $1665\mu s$ respectively. However, when two packets are aggregated, it takes only $846\mu s$, which accounts for about 50% time saving. The value is got by adding the time contribution of additional 20 bytes payload $(14.5\mu s)$ to the normal transfer time for one *VoIP* packet $(832.5\mu s)$. Thus, only a small number of *VoIP* packets can be supported in *WMNs* since protocol overheads take a good portion of the bandwidth.



(a) Time to transfer un-aggregated packets



(b) Time to transfer aggregated packets

Fig. 3. Aggregation of two packets

When aggregating, an extra overhead of *20ms* is usually added to the first packet. This makes it illogical aggregate in lightly loaded networks. However, under a heavily loaded network, which usually happens in *WMNs*, the small packets experience heavy contention. The increased contention causes voice packets to drop or be retransmitted resulting into increased network traffic. In such networks, packets have to be queued while waiting for media access. The packets in the queue form good candidates for aggregation as the queuing time can be used to aggregate. To illustrate the benefit of packet aggregation, assume that packets of the same size ρ bytes are transmitted at a channel rate of λ*Mbps*. The benefit of aggregating κ packets during transmission can be determined by getting the difference between transmissions with aggregation and without aggregation. The saved time τ seconds, can thus be expressed as follows.

$$\tau = \tau_0.(\kappa - 1) - \frac{8.\gamma}{\lambda}, \qquad (2)$$

where γ denotes the size of aggregation header and τ_0 is the channel time. Since γ and λ may be assumed constant for IEEE 802.11b based *WMNs*, by inspecting (2), it can be noted that the aggregation benefit τ increases with increase in the number of packets. Although this implies that “the larger the aggregation size the better”, the implementation prompts for further considerations on end-to-end delay, delay variance and packet loss parameters which are crucial for quality *VoIP*. This paper focuses on the performance of *VoIP* over *WMNs* under no aggregation, *fixed* aggregation, and the proposed dynamic aggregation.

A. Fixed Aggregation Algorithm

This is also called *forced-delay* aggregation algorithm. The algorithm marks arriving packets with a timestamp. The marked packets may then be delayed for a pre-defined time called maximum wait period (δ seconds). After the expiry of δ, packets destined to same next hop are aggregated. The size of the aggregated packet is however limited by the maximum transmission unit (*MTU*), which is no more than 2300 bytes for IEEE 802.11 standard [12]. The right choice of δ is important. Higher values for δ yield higher aggregation rate, but also a higher end-to-end delay. Packets get aggregated only when they reach a defined size. Although advantageous in areas where traffic is heavy, lager packets generated due to aggregation may lead to higher packet loss in erroneous channels.

In this work, the fixed aggregation uses a maximum packet size of 1500 bytes and a maximum delay time of 6ms. Packet aggregations at the intermediate nodes uses queuing delay and only induce forced delay when *MAC* is not busy.

B. Dynamic Aggregation Algorithm (DA)

The DA is similar to fixed aggregation algorithm except that it uses local link characteristics to determine adopt an appropriate packet size that reduces chances of packet loss. This packet size is called aggregation threshold and is not more than *MTU*.

The decision on when to aggregate is influenced by two parameters: maximum queue size φ_l, and delay time χ_l. If a link has a queue size greater than φ_l or a head-of-line packet timestamp indicates it is χ_l old, then the packets in the queue are aggregated. During this time, *VoIP* packets are packed together until the size of the new packet becomes larger than aggregation threshold or the queue becomes empty. If no queue satisfies the conditions, the node stays idle. This releases the wireless channel to be used by other nodes. The two parameters, φ_l and χ_l, are related by

$$\varphi_l = \beta . \chi_l , \qquad (3)$$

where β, is the average input rate of link l. When l is given, the primary problem is to determine how to choose χ_l for each wireless link. The packet aggregation rate of link l is defined as

$$\psi_l \equiv 1/\chi_l . \qquad (4)$$

Here, the optimal value of (4) minimizes packet delay in *WMNs*. However, the optimal value for ψ is constrained by flow conservation (*FC*), capacity limit (*CL*) and *MTU* size properties. The *FC* property emphasizes that the incoming data rate of a link is equal to the outgoing data rate. This data rate is also the aggregation rate. The capacity constraint ensures that the utilized capacity is no more than the capacity that the channel can offer. As for the *MTU* size, the aggregated packet size should not exceed *MTU.*

Since aggregation aims at achieving higher capacity by combining smaller packets, the packet rate formulation narrows down to determining the maximum packet size that optimizes (4). Besides, for a given channel quality, contention level and traffic injection rate, different packet sizes produce different packet loss ratios. Packet loss in *WMNs* is dependent on the bit error, queue overflows, and collisions. Packet loss due to collision and queue overflows can be reduced by increasing packet sizes. However, larger packets increase packet loss due to bit error.

Bit error occurs when a received signal cannot be decoded properly. The extent of bit error called bit error rate *(BER)* is dependent on the modulation scheme, signal-to-noise and Interference ratio (*SNIR*) of the received signal, the coding scheme and data rate [13]. Here, apart from *SNIR,* other factors are usually constant in IEEE 802.11b based networks. The *BER* is therefore only dependent on *SNIR.* According to [14], *SNIR* is defined as follows.

$$SNIR = 10 \log_{10}\left(\frac{P_s}{P_n}\right), \qquad (5)$$

where P_s is the strength of the signal and P_n is the strength of noise produced by thermal noise and interference. Therefore, by defining the following variables: $D_i = (1-\alpha(\beta,R_i))^{L_i}$, $D_j = (1-\alpha(\beta,R_j))^{L_j}$ and $D_k = (1-\alpha(\beta,R_k))^{8.L_k}$, a relationship between frame error rate (*FER*) and *BER* may be expressed as follows [14].

$$FER = 1 - D_i . D_j . D_k , \qquad (6)$$

where, α is the *BER*, β is the *SNIR* value, R_j is the transmission rate of preamble, R_i is the transmission rate of physical layer control protocol (*PLCP*) header, R_k is the transmission rate of *MAC* frame, L_j is the length of the preamble bits, L_i is the length of *PCLP* header in bits and L_k is the length of *MAC* frame in bytes.

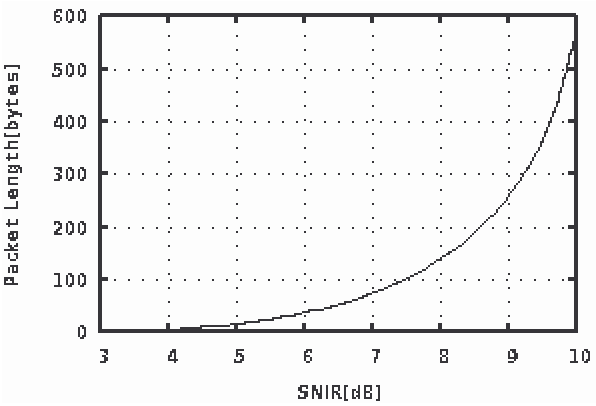


Fig. 4. Correct Packet length for a given SNIR [15]

If the lengths of the preamble and header, and transmission rates are considered to be constant, the *FER* is a function of *SNIR* and the frame length. For any network, as the *SNIR* goes to infinity the average error rate goes to zero. This means that the network becomes more accommodative to larger packets as the *SNIR* gets higher. Fig. 4 illustrates the relationship between packet size and *SNIR* assuming IEEE 802.11 standard overheads. With these arguments, an optimal packet determination scheme can be developed as a function of *SNIR.* The scheme incorporates the handshake between the sender and the receiver. The receiving node measures the *SNIR* of the coming packets, calculates the maximum tolerable packet size based on the current *SNIR* and transmits the calculated value to the sender. The current *SNIR* value (S_{k+1}) for each link is calculated and stored in the routing table. The formula used is

$$S_{k+1} = S_k + \sigma(S_m - S_k)\ , \tag{7}$$

where S_k defines $SNIR$ value before receiving the current packet, S_m is the $SNIR$ of the incoming packet and σ is the smoothing factor. Since static *WMNs* are stable, the value of σ should be as low as 0.1.

V. PERFORMANCE EVALUATION

This section evaluates performance of the *DA* in terms of end-to-end delay, jitter and packet loss of VoIP packets. The results are compared with those recorded under no aggregation and *fixed* aggregation approaches. Simulations are done in ns-2 under the settings provided in Table I. During simulations the number of concurrent flows is varied so as to model different degrees of network contention and interference. This aids in understanding the performance of the *DA* over real mesh network deployments.

TABLE I
SIMULATION SETTINGS

Parameter	Value
Simulation Time	150 seconds
Propagation Model	Log-normal Shadowing
Path Loss Exponent	2.5dB
Transmit Power	15dBm
Frequency	2.4e+9
RXThreshold	-71dBm
CSThreshold	-109dBm
Path Loss Exponent	2.5dB
Data Rate	11Mbps
Basic Rate	2Mbps
PLCP Preamble	Short Preamble
MTU	1500 bytes
RTS/CTS	Off
Queue Length	100
MAC Access Mechanism	DCF
MAC Layer Protocol	IEEE 802.11b
Routing Algorithm	AODV-UU
Aggregation Delay	6 milliseconds

Ns-2 does not come with a *VoIP* traffic agent. In this paper, *VoIP* is modelled as a bidirectional flow with silence suppression as an on-off Markov process. The conversation is assigned a talk spurt of 35% and silent periods of 65% as typical with G.729A vocoder. *VoIP* is transmitted over UDP/RTP/IP protocols to form a total packet size of 60 bytes

Fig. 5 illustrates the network topology used in this work. It comprises of wired and wireless mesh clients, an *AP* to provide access to the Internet and wireless mesh routers to extend the coverage of *APs*. This arrangement of nodes replicates the current single radio networks where the closest gateway is usually no more than two hops. The network assumes that there is only one *AP* in the network. Nodes in the network are configured for hierarchical routing and assigned static IP addresses.

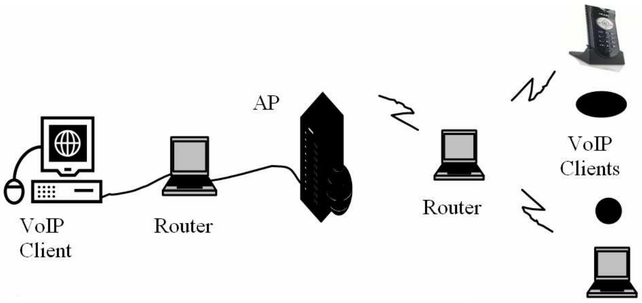


Fig. 5. Simulation topology

Fig. 6 illustrates the end-to-end delay characteristics for three scenarios. Looking at the figure, it can be seen that for low traffic, aggregation algorithms have higher traffic delay as compared to no aggregation. However, as the number of injected flows increases, more packets get aggregated and thus reducing the average packet delay. The *DA* has superior performances with a brink experienced from 105 flows compared to 45 and 30 for fixed and no aggregation.

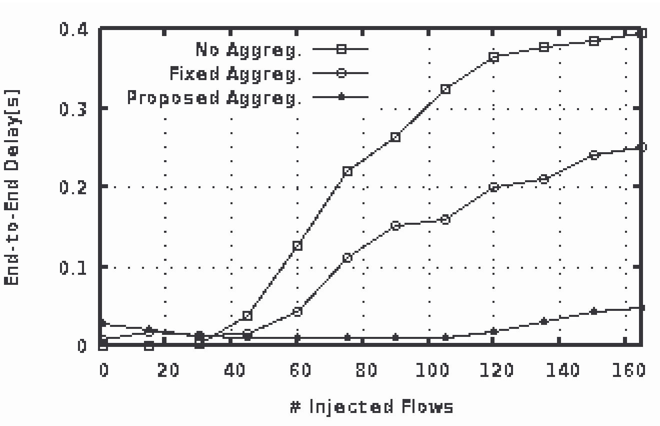


Fig. 6. End-to-end delay for VoIP in WMNs

In Fig. 7, the experienced jitter is plotted against injected flows. From the figure, it can be seen that packet aggregation reduces delay variation. By sending large blocks of packets, aggregation algorithms reduce chances of having unnecessarily longer queues that causes jitter in the network. The results shows that the *DA* experiences a brink after 105 flows while fixed aggregation and no aggregation have their jitter rising from 30 and 25 flows respectively.

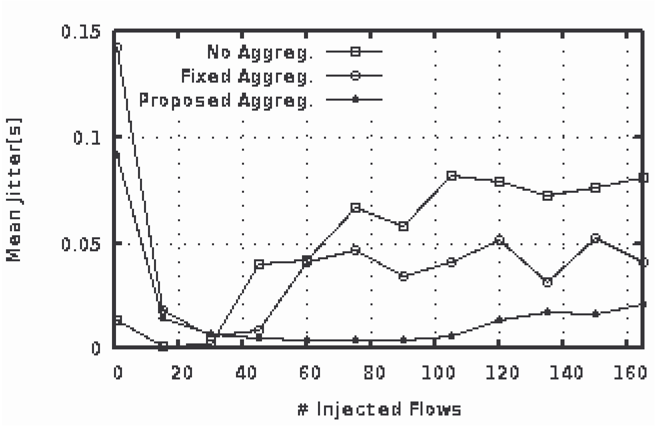


Fig. 7. Average delay variation for VoIP packets

However, from Figures 6 and 7 it can be seen that it is better to send packets without aggregation. The inferior performance of aggregation here occurs since for lower traffic some packets are delayed due to the δ delay parameter and queuing. As a result packets require different time to be transferred. If δ is small, most packets are sent without aggregation thus demystifying the use of aggregation.

Packet loss rate is also a crucial parameter in evaluating network performance. It includes both packets that do not reach the destination within the required time. The lager packets generated due to aggregation have higher chances of being dropped due to frame errors conditions. As seen in Fig. 8, fixed aggregation that uses constant aggregation packet size experiences larger packet loss compared to other techniques. The use of no aggregation experiences higher packet loss as a result of jitter buffer being overwhelmed by large number of packets

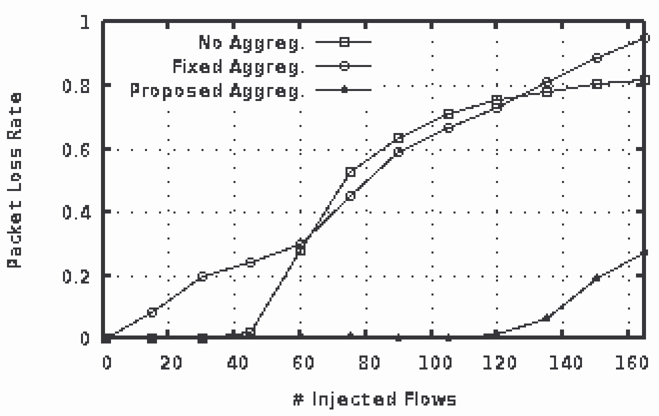


Figure 8: VoIP packet loss rate in WMNs

The better performance realized by the *DA* is attributed to the ability of the algorithm to vary packet size in response to link characteristics. The fixed aggregation algorithm may create packets that are too large to be accommodated in a channel leading to a drop to packet loss. However, even below the capacity threshold it happens that some flows have bad quality. Ideally all flows below threshold are to be supported and this divergence can only be attributed to the difference in confidence levels between injected flows

VI. CONCLUSIONS

This paper proposed a link based aggregation algorithm that dynamically determines the acceptable aggregation packet size based on local link characteristics. The algorithm is simulated and its performances compared with no aggregation and fixed aggregation approaches. The simulation results show that the *DA* has favourable end-to-end delay, jitter and packet loss guarantees compared to other approaches for a given number of parallel flows up to a threshold limit. Thus, the algorithm proves that link characteristics can be exploited to ensure quality *VoIP* transmission over *WMNs*.

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James M. Okech, Kenya, 1979. Received the B.S in Computer Engineering degree from Maseno University, Maseno, Kenya in 2002 and is currently working towards M.T degree in Telecommunication Technology and M.S degree in Electronics Engineering at the Tshwane University of Technology, Pretoria, South Africa.

From 2003 to 2004 he was associated with Oshwal College, Nairobi, Kenya as an Assistant Lecturer in the Computer Department. Between 2004 and 2007 he was with Maseno University, Maseno, Kenya as a Graduate Assistant in Computer Department. His research interests include networking, network performance analysis and protocol design for multimedia applications over wireless mesh networks.