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 $O(n^{-\alpha})$, 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.
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 14.5μs at 11Mbps, but IP/UDP/RTP header, MAC headers and physical headers, inter-
frame periods, back-off and acknowledgements (ACK) need a total of 818μs. The contribution of the VoIP payload leads to a cumulative transmission time of 832.5μs. This is about 12 VoIP calls supported per hop. The number of supported calls is calculated using:

\[(2.\beta\alpha)^{-1}, \]

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.

![Packet Aggregation](image)

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 832.5μs and 1665μs respectively. However, when two packets are aggregated, it takes only 846μs, which accounts for about 50% time saving. The value is got by adding the time contribution of additional 20 bytes payload (14.5μs) to the normal transfer time for one VoIP packet (832.5μs).

Thus, only a small number of VoIP packets can be supported in WMNs since protocol overheads take a good portion of the bandwidth.

![Aggregation of two packets](image)

(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 p bytes are transmitted at a channel rate of 1Mbps. The benefit of aggregating k packets during transmission can be determined by getting the difference between transmissions with aggregation and without aggregation. The saved time \(\tau\) seconds, can thus be expressed as follows.

\[\tau = \tau_0 (k-1) \frac{8\gamma}{\lambda},\]

where \(\gamma\) denotes the size of aggregation header and \(\tau_0\) is the channel time. Since \(\gamma\) and \(\lambda\) may be assumed constant for IEEE 802.11b based WMNs, by inspecting (2), it can be noted that the aggregation benefit \(\tau\) 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 (\(\delta\) seconds). After the expiry of \(\delta\), 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 $\delta$ is important. Higher values for $\delta$ 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, larger 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 $\phi_l$ and delay time $\chi_l$. If a link has a queue size greater than $\phi_l$ or a head-of-line packet timestamp indicates it is $\chi_l$ 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, $\phi_l$ and $\chi_l$, are related by

$$\phi_l = \beta \chi_l,$$  

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

$$\psi_l = \frac{1}{\chi_l}.$$  

Here, the optimal value of (4) minimizes packet delay in WMNs. However, the optimal value for $\psi$ 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.

$$SNIR = 10 \log \left( \frac{P_s}{P_n} \right),$$  

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: $D_1 = (1-\alpha(\beta, R))^L_s$, $D_2 = (1-\alpha(\beta, R))^L$, and $D_3 = (1-\alpha(\beta, R))^8L_s$, a relationship between frame error rate (FER) and BER may be expressed as follows [14].

$$FER = 1 - D_1D_2D_3,$$  

where, $\alpha$ is the BER, $\beta$ is the SNIR value, $R_s$ is the transmission rate of preamble, $R_i$ is the transmission rate of physical layer control protocol (PLCP) header, $R_j$ is the transmission rate of MAC frame, $L_s$ is the length of the preamble bits, $L_i$ is the length of PLCP header in bits and $L_j$ is the length of MAC frame in bytes.

Fig. 4 illustrates the relationship between packet size and SNIR assuming IEEE 802.11 standard overheard. 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 ($S_{new}$) for each
link is calculated and stored in the routing table. The formula used is

$$S_{i+1} = S_i + \sigma(S_m - S_i)$$ \hspace{1cm} (7)

where $S_i$ defines SNIR value before receiving the current packet, $S_m$ is the SNIR of the incoming packet and $\sigma$ is the smoothing factor. Since static WMNs are stable, the value of $\sigma$ 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

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Simulation Time</td>
<td>150 seconds</td>
</tr>
<tr>
<td>Propagation Model</td>
<td>Log-normal Shadowing</td>
</tr>
<tr>
<td>Path Loss Exponent</td>
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</tr>
<tr>
<td>Transmit Power</td>
<td>15dBm</td>
</tr>
<tr>
<td>Frequency</td>
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<td>RXThreshold</td>
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</tr>
<tr>
<td>CSThreshold</td>
<td>-109dBm</td>
</tr>
<tr>
<td>Path Loss Exponent</td>
<td>2.5dB</td>
</tr>
<tr>
<td>Data Rate</td>
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</tr>
<tr>
<td>Basic Rate</td>
<td>2Mbps</td>
</tr>
<tr>
<td>PLC Preamble</td>
<td>Short Preamble</td>
</tr>
<tr>
<td>MTU</td>
<td>1500 bytes</td>
</tr>
<tr>
<td>RTS/CTS</td>
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</tr>
<tr>
<td>Queue Length</td>
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</tr>
<tr>
<td>MAC Access Mechanism</td>
<td>DCF</td>
</tr>
<tr>
<td>MAC Layer Protocol</td>
<td>IEEE 802.11b</td>
</tr>
<tr>
<td>Routing Algorithm</td>
<td>AODV-UU</td>
</tr>
<tr>
<td>Aggregation Delay</td>
<td>6 milliseconds</td>
</tr>
</tbody>
</table>

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. 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.

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.
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 $\delta$ delay parameter and queuing. As a result packets require different time to be transferred. If $\delta$ 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 larger 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.

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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